An Adaptive Technique for the Acoustic Transmission of Digital Data from Subocean Platforms

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The work described herein was undertaken on behalf of the U.S. Geological Survey's R&D Program for OCS Oil and Gas Operations.

It is an attempt to advance the technology for the underwater acoustic telemetry of data which is required during various offshore operational situations. Work is continuing.

If you wish to comment on this report, or desire additional information, please contact either the principal investigator, Dr. Softley or Mr John B. Gregory, Research Program Manager.

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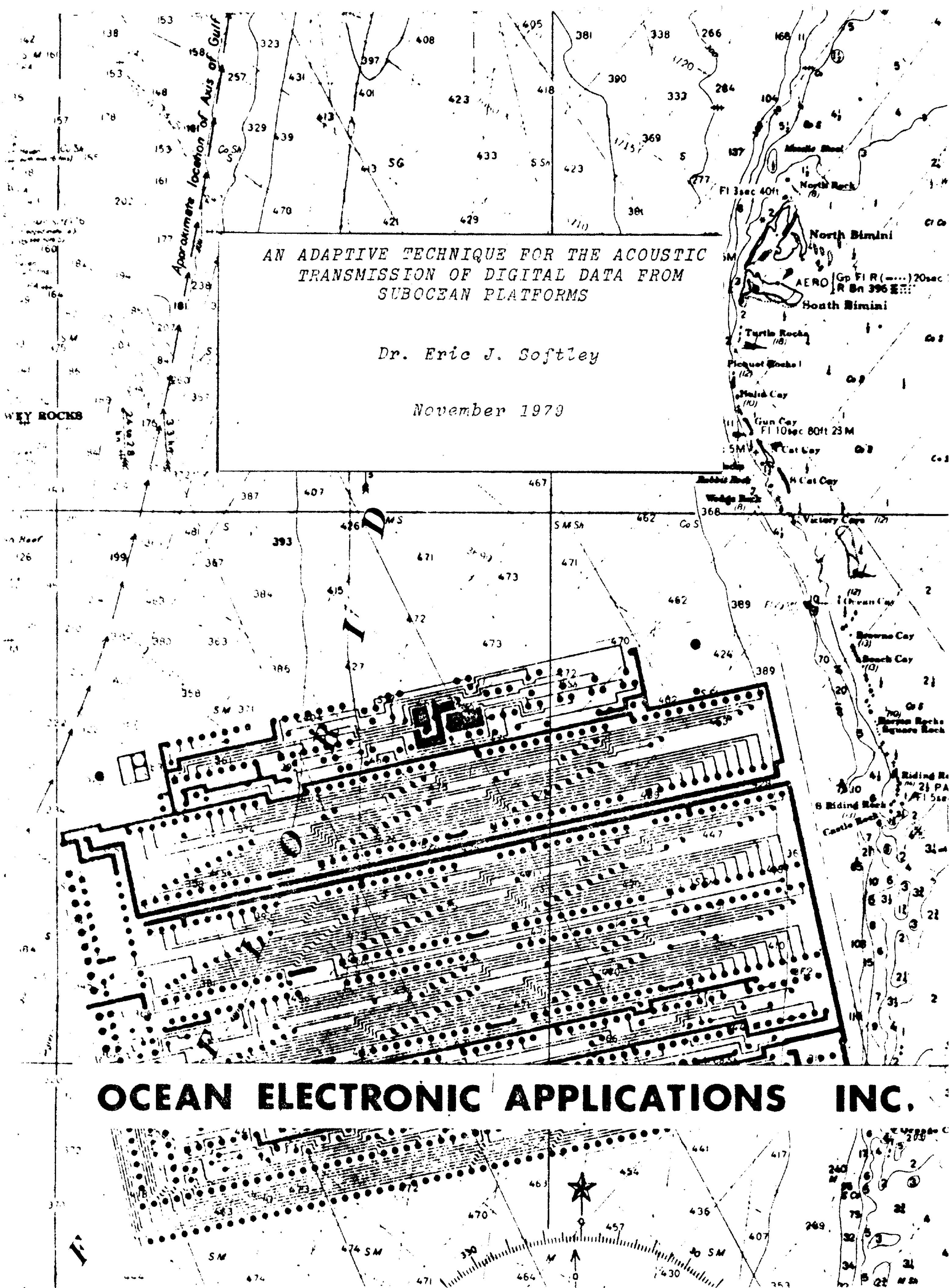

OCEAN ELECTRONIC APPLICATIONS INC.

AN ADAPTIVE TECHNIQUE FOR THE ACOUSTIC TRANSMISSION OF DIGITAL DATA FROM SUBOCEAN PLATFORMS

Dr. Eric J. Softley

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1. INTRODUCTION

As the offshore industry continues to move into deeper and more hazardous frontiers, the need for underwater information increases. Cable systems, currently used extensively to obtain data from unmanned facilities, are cumbersome to handle and control and are expensive because of the length of cables needed. In addition because they reach upwards through the water column they are especially subject to the rigors of the sea and are a major source of system failure. Moreover, the advent of modern electronic components has resulted in a reduction in the size, weight and cost of data gathering systems, making them less and less compatible with the cables commonly in use.

It is apparent from our knowledge of acoustic propagation in the ocean that the possibilities exist to interrogate underwater packages acoustically and to obtain data from them. In fact many investigators and developers of technology have done just that[1,2,3]. In general, however, these systems are fixed in terms of their operating parameters and find application to very specific geographies and ocean topographies. If a more general problem is considered a number of problems immediately become apparent. These have to do with the propagation of acoustics in the water and in particular the effect of variations of the ocean parameters on the acoustic signal received. As an example we will consider three scenarios as typical of USGS operational needs and examine the important parameters associated with them. The three scenarios selected are:

(1) the nondestructive testing of large ocean towers;

(2) operational support of unmanned, untethered vehicles and;

(3) interrogation of underwater instruments by use of air launched acoustic transceivers.

These three scenarios do not represent all those of interest, however, they do demonstrate the wide variation in parameters important to data propagation.

The important parameters for the three scenarios are shown in Table I. Common to all three are several basic factors. First the water depth would not, in general, exceed 300 meters. One primary difference is in the horizontal range expected. In the case of a tower with instrumentation on the lower legs (underwater) the path length would be essentially the vertical height of *the water and indeed the path traveled would be essentially vertical. However, in the case of the unmanned vehicle, which is expected to be mobile, this could conceivably reach over an eight hour period a distance of five kilometers or more away from the monitoring surface vessel. In the case of the air deployed packages navigational accuracy of the aircraft would result in a range of two kilometers or more.*

The data to be transmitted is quite varied. The nondestructive testing technique essentially generates a batch of data over a short period. If we assume that this is stored and transmitted at a reduced rate the actual total data rate requirements are quite modest. They might, for example, involve 10^5 bits of data. However, the testing would not be done at a rapid rate and this data could be transmitted at a fairly low rate, as low as 100 baud if necessary.

The vehicle represents a different problem in that a number of possibilities for data transfer can take place. Control of the operation of the vehicle, essentially a secondary function since the vehicle is assumed to have an onboard control capability, involves only small amounts of data, indeed something on the order of 10^3 bits of data, transmitted at a hundred baud rate would

This lifetime reflects seriously in the amount of energy that could be carried. For example, using lithium batteries with a one megajoule per kilogram capability a total of one megajoule typically would be available over a one year period for the first and third scenarios. However, in the case of the vehicle with possibly rechargeable batteries something like 500 kilojoules of energy would be available but this could be extended over a fairly short period of time. A basic common denominator of all three systems and one which is common to all underwater operations is that energy needs to be conserved and used as sparingly as possible.

If the above scenarios and data needs are compared with the acoustic telemetry characteristics several interesting factors occur. In general the higher the acoustic frequency the more rapid its attenuation due to absorption. Therefore the longer ranges would tend to use lower acoustic frequencies while the shorter ranges could use higher acoustic frequencies. The matter gets somewhat more complicated if a basic problem concerning acoustic propagation in shallow water is considered, namely that of multipath interference. Here in addition to a primary acoustic path other acoustic paths, generally involving reflections from the surface of the ocean or from the bottom, also cause signals to arrive at the receiver. Because these paths lengths are different, then the signal generated arrives at different times causing interference with the signal and an increase in the background noise. Here a higher acoustic frequency would in general give more attenuation of the second and third paths and the multipath problem would be reduced. However, the primary signal would also be reduced to the point that the background noise in the ocean would be a factor. An increase in the transmitted power would help at the expense of more rapid depletion of the energy source.

Another problem that occurs is that temperature and salinity fluctuation in the ocean cause temporal fluctuations of the arrival time of the signal at the receiver. This results in jitter of the data bits with the implication that the data rate may depend on local environmental fluctuations. In some experiments performed by the author several years ago we were able to transmit data satisfactorily a distance of two kilometers with fairly low transmitted power (of one watt) for some of the time. However, at other times the data jitter was sufficient that no data could be coherently received with a fixed system.

It becomes apparent that to create an acoustic data telemetry system capable of satisfying a number of different scenarios in different oceanographic locations and at different times, the various acoustic parameters of importance should be variable. These parameters are the acoustic frequency, the power level and the data rate. These variations of primary parameters would have to occur as a result of measurements made during the data transmission. Thus the system would adapt to the environment adjusting the parameters to optimize the link performance.

Examining the feasibility of this adaptive system is the subject of the present study. The first year of experimentation and development has involved the concept development and hardware development of a system capable of performing the adaptive process. (During the next year we plan to develop the software and the adaptive logic).

2. *SYSTEM CONCEPT*

2.1 *Acoustic Considerations*

The adaptive system discussed in the introduction has acoustic frequency, power level and data rate as variables. During any transmission these parameters are determined from analysis of the received signal. In many cases a preliminary conversation between stations will be needed to optimize the parameters. This will be discussed in detail later.

Adjustment of the acoustic frequency is made primarily to account for a varying path length. Some idea of the effect of change in frequency can be seen in Figure 1 where the effects of spherical spreading and absorption loss for different frequencies are considered as a function of range. If we make a basic assumption that the frequency to be transmitted would be as high as possible, to allow the highest bit rate, then it could be seen that the choice of frequency would depend on the background noise. The strength of the received signal is shown, assuming an 170 dB (re 1 μPa @ 1M) source in all cases, for the eight channels now being used in the system. Table II lists these channels and shows the center frequency of transmission and the theoretical maximum range assuming that the signal to be received is 6 dB *above "typical" coastal noise[5]. The range varies significantly with acoustic frequency. With upper limit of 10 kM (defined in Table I) we can see that a frequency as low as 14 kHz would allow this range. For shorter path lengths higher frequency channels could be used. These selected acoustic frequencies vary over a range of 14 kHz to approximately 90 kHz. At frequencies significantly above 90 kHz the background noise in the ocean tends to increase and it also becomes more difficult to generate the signals with the same transducers. For convenience eight channels are defined, each increasing frequency representing a 30% increase over the next lower channel.*

The frequencies will be modulated using the data so that the actual transmitted frequencies are $\pm$ 9% of the center frequency. This deviation is digitally generated in the system. This modulation allows the use of a fast attack automatic gain control (AGC) in the receiver. Figure 1 shows that the received signal will vary from $\pm$ 40 dB to over 100 dB (re 1 μPa). Hence a second requirement of the system is that it have an AGC with 60 dB range. Greater AGC control is not necessary because in addition to the variations of frequency 18 dB output power variation is employed allowing a reduction of power at short ranges, for example, less than 1 kM.

The data rates shown in Table II are maximum values from theoretical analysis of the bit discriminator. These are in fact a direct result of the choice of the type of coding used. Clearly because of the variable bit rate plus the natural bit jitter expected from acoustic transmissions some kind of coding is required that is self-clocking. The common used method of coding is Manchester or split phased modulation (SPM) which allows the recreation of the bit rate from transitions within the data. SPM requires a band width of approximately twice the bit rate. However, one disadvantage of using Manchester is that the bit clock occurs on both positive and negative transitions and in fact there is an ambiguity in the data. Since the receiver will not know the acoustic frequency or any of the other parameters it will be necessary to scan the signal looking for the channel and determining the bit rate during the initial transmission. For this reason a form of clocking which does not contain any ambiguity is needed.

The modulation chosen is pulse duration modulation (PDM) where the bit clock always occurs on positive transitions of the data and the data itself is determined from the location in time of the negative transition. Figure 4 demonstrates this by comparing PDM and SPM for a typical signal. The maximum bit rate shown in Table II derives naturally from the use of PDM from the fact that the bits are not synchronous with the acoustic signal and therefore there is an ambiguity of one cycle in recognizing a high or low level.

In practice the maximum bit rate would be expected only under very ideal conditions. As mentioned earlier several experiments by the author in the past had indicated that significant bit jitter occured due to fluctuations in the ocean. This adds to the difficulty of coherently converting the data from serial to parallel form. Actual bit rates used by the system are shown in Table III. It can be seen that the bit rate varies from a value close to the theoretical limit down roughly a factor of eight.

Hence the system has a fixed modulation type, fixed digital coding, variable acoustic frequency over a factor of eight and an output power variable over some 18 dB. It is intended that the system be adaptable to data transfer from close ranges in the order of 50 M to long ranges in the order of 10 kM.

It should be observed that the problem of multipath signal interference is not solved by varying the parameters. However, adjustment of the acoustic frequency to the highest possible under the varying circumstances does allow this problem to be minimized. In fact multipath discrimination is an inherent key to the design of the receiver signal decoder as will be seen later.

As has been observed earlier, that with the basic acoustic parameters as variables then the receiver has to have the ability to evaluate the incoming signal and to decode accordingly. Basic receiver requirements are the ability to measure the background noise, to measure the signal level, to determine that sufficient signal excess exists, to measure the multipath signals level and to determine that they are sufficiently low, to measure bit rate and to determine that the bit jitter is within tolerable limits. This hardware will be described later.

It should be noted that the data idealized in Figure 1 assumes that the source level is constant for all frequencies. While it is relatively simple to produce a power amplifier which is flat over the useable spectrum it is not necessarily simple to produce an acoustic transducer that would reproduce the signal with a flat response. Early in the program an attempt was made to purchase a transducer capable of operating over the frequency range shown in Table I. However, the cost and in particular the delivery time of a suitable transducer, for both transmitting and receiving precluded the purchasing of this device at this time. Therefore the experiments to be described later were performed using a transducer that was created in house. A description is given later in this report. It should be observed that this transducer was used for both transmitting and receiving and a simple calibration of the transfer function, i.e. the product of the transmitting response and receiving response was made. This indicated a transducer peaking at 30 kHz but useful down to 15 kHz and up to over 100 kHz. While the response in the transfer function certainly was not flat it was useful for the first experiments. A combination of transducers to give us a reasonably flat response over the frequency range could be employed at a later date.

2.2 The System Layout and Function

Two transceivers, essentially identical, are needed, see Figure 2. One of the two transceivers is a submarine package housed in a suitable pressure housing. This device would accumulate data and transmit on request. The second package, located typically on board ship, would have a hydrophone or hydrophone array suspended over the side and transmit requests for data and receive data. During the initial experiments the submarine package was used only to transmit sample data at various acoustic frequencies, bit rates and power levels. The shipboard package was used only in the receive mode to receive and evaluate the data. During these experiments we were evaluating aspects of the receiver and in particular the acoustic signal decoder. In future experiments, during the coming year, it is planned to complete both packages as full transceivers, to operate them in an adaptive mode and to develop the procedure for making adjustment of the acoustic parameters.

The system layout for each of the transceivers is shown in block form in Figure 4. Each transceiver consists of two microcomputers. The first computer is a data computer. It accumulates data and handles the problems of data formatting and initiation of the transmission. It generates the coded serial data which it sends to the acoustic computer. It also sends a control word which defines transmission frequency and power level. In the transmitting mode the acoustic computer then generates and transmits the acoustic signal. In the receive mode the signal enters the acoustic computer which detects the level of the incoming signal and uses a programmable *gain amplifier to digitally control the gain of the signal to 0 ± 0.3 dB. The gain of the input signal is read directly by the data computer and hence the level of the incoming signal is derived directly. The acoustic computer inputs the constant level signal to a set of digital filters and to a data discriminator which outputs serial digital data.*

The operational procedure of the data computer in the receive mode has several stages. When first initialized or immediately following a transmission the data computer performs a background noise scan. Each of the channels is selected in turn, the acoustic computer initiated and the gain of the incoming signal is determined. This is defined as the background noise level at that time for each channel. It should be observed that a signal could be present at this time. However, this means than an initially false background level would be determined and this would be upgraded immediately.

Following the prescan the data computer then performs a continous scan of the eight channels. Each channel is selected, there is a twenty millisecond pause to enable the automatic gain control to settle to a stable level and the gain is recorded. After all eight channels are recorded the data computer evaluates each channel signal level to determine whether any signal is above the background signal by sufficient margin. If not then the background data is updated.

If a signal is present then this channel is selected and the incoming serial data examined to determine if a coherent bit duration exists. This is done by taking four successive bits, measuring their duration and comparing them to a reference error level to determine if transmissions are suitably coherent with time. Assuming bit coherence exists then the data computer will examine sequential bits to determine the duration of the one bit and the duration of the zero bit.

The computer then continually examines the incoming data stream for the sync word that indicates the initiation of the transmission. The first sixteen words of any transmission contain the channel word in a highly redundant and error correcting form. The data computer will examine this to determine the bit error rate. It is recognized that in 256 sequential bits low bit error rates will not necessarily show up. In terms of evaluating the consistency of the channel as a data channel all 256 bits must prove valid in order for the data computer to consider subsequent data transmissions as valid.

In addition to being able to evaluate the background noise the signal level relative to the background noise and the coherence of the incoming bit data, the data computer can also evaluate from the transverse filter elements the noise level within a narrow signal band. This noise level, if significantly above the background noise, would indicate the presence of multipath signals arriving and the computer can measure these relative to the incoming signal to determine whether the multipath signal is strong.

2.3 System Operation

Thus the data computer would have all information available to it concerning the received signal characteristics that determine the quality of the received signal as well as information on the transmitted signal. It can then proceed to make decisions as to whether the channel, power level and bit rate are correct for continued data transfer or whether the channel should be changed to improve the level of signal, to improve the bit rate or to decrease the power.

It is apparent that when communication is first established the first choice of parameters may be far from optimum. Hence initially the 256 bit message, known as the calibrator frame, which precedes all transmissions are transmitted back and forth. As discussed earlier it contains a highly redundant form of the channel word (CW). This channel word consists of 12 bits of data which describe the transmission. Table IV illustrates the channel word. In addition to the channel number, power level and bit rate, it also contains a two bit word indicating the message status (MS).

Initially the requesting system, usually located on board ship, will send a calibration frame with MS = 1, requesting data but with unevaluated acoustic parameters (CW). Assuming that these parameters are not ideal the submarine system will reply with a new set of parameters and MS = 2. This iteration proceeds until one or other of the systems determine adequacy of the communication. If the ship system determines adequacy the next calibration frame has MS = 3. If the submarine determines adequacy, or if MS = 3 in the received frame, then the requested data is transmitted with MS = 0 and the process is terminated.

At first glance this process sounds lengthy. However, each transmission takes typically 1¼ seconds consisting of 1 second of bit clock and 250 milliseconds of data. (The 1 second bit clock is used to allow time for the receiver to acquire the channel.) Since the channel evaluation computation is quite quick six iterations on the calibration frame can be performed in under ten seconds as an example of the total adaptive process.

3. *HARDWARE DEVELOPMENT*

3.1 *Package Description*

As discussed in the earlier section acoustic communication between a submarine and a ship mounted package would require two transceivers each capable of receiving and transmitting. Figure 3 shows the basic breakdown of each transceiver into two microcomputer sub-assemblies. Within each of these components, sections of each computer are used in transmitting and receiving. Figure 5 shows a more detailed breakdown of the transceivers. The external data is accumulated by the data computer and stored in a mass memory which at present time is designed for 10^6 bits of data. On transmission the data is formatted in a coder section and a suitable control word initiating the acoustic transceiver is applied to the control bus. The acoustic computer interprets the control word and commands a synthesizer which generates frequencies pertinent to the particular channel. The function generator uses these frequencies in conjunction with the data out to generate a frequency modulated signal. The output of the function generator is amplified and the signal sent through a transmit relay to a transducer.

During receiver operation signals from the transducer enter via the relay to a low noise preamp whose output enters a controllable analog filter. This analog filter serves to reduce the background noise and improve the signal to noise ratio of the received signal. This signal enters the automatic gain control, consisting of a programmable gain amplifier controlled by the control processor and a constant level signal, outputs from the AGC to a set of digital filters. The output of these filters is detected by a presence detector which regenerates the data input stream and sends it to the data decoder which allows the 6100 to interpret it.

It should be observed at this time that the two computers each uses its own bus and a third control bus interconnects the two. CMOS devices are used extensively. The data computer, for example, is 100% CMOS. Within the acoustic microcomputer the power amplifier, analog filters and digital filters are the only non-CMOS devices. It could be noted that the data computer consumes less than 20 milliamps during data receipt. These are very small compared to the power consumed by the power amplifier on transmission.

During the first year of experiments two partial transceivers were built. The basic functions used in transmission were incorporated in one device; the second device was completed as a receiver. The trans-

mitted device was used as a submarine package to transmit known data and the receiver was located on board ship to enable the receiver to be analysed and to examine the data input. Current work involves completing both transceivers as shown in Figure 5 so that two way data may be examined.

3.2 *The Data Computer*

The data computer is based on an Intersil 6100 microprocessor. This microprocessor which emulates the well known Digital Equipment Corporation PDP 8E computer and uses the PDP 8 instruction set is a CMOS device with all advantages of CMOS architecture. A variable speed microprocessor clock allows adjustment of power as computing needs vary and power is disabled for periods of time when computing is not necessary. The 6100 microprocessor is the basis for an all CMOS data computer. This computer was developed for the ADOM program[4] and parts of it form the basis for the hardware used in the experiments of this program.

Other elements of the data computer are a PROM memory which contains the operational program in nonerasable form and an extensive RAM memory which can be as small as 1K words (each word is 12 bits) or directly address up to 2048K words of memory. This directly *addressable memory is some 64 times larger than the maximum PDP 8 memory. Each 32K word memory segment forms a complete PDP 8 equivalent and can be operated as such. The extensive additional memory allows the cumulation of data up to a maximum of 2.7×10^7 bits and this is felt to be adequate for most of the data requirements envisaged. The present building block for this memory is the 4K CMOS static RAM and each module of the computer contains 8K x 12 bits of data or approximately 10^5 bits. For the full memory capability the number of boards required would be very large. However, the basic memory scheme can use other devices as they become available and the number of boards required for such a large scale memory will reduce accordingly. The present hardware contains 12 boards, each of 10^5 bits.*

Other components of the data computer include an interface board to control the acoustic computer, read the gain of the acoustic system and to code output data. This coder has been designed so that both split phase modulation (SPM) and pulse duration modulation (PDM) can be generated. Only pulse duration modulation has been used in the tests. The module which contains the interface control and the data coder is shown in somewhat more detailed form in Figure 6. The data is loaded into a first *in first out (FIFO) array which allows the storage of 16 words, 12 bits wide. This is turned into NRZ serial data and then through a modulator which generates SPM or PDM. An external clock (from the acoustic computer and proportional to the basic frequency of the synthesizer) is divided as required to produce the correct baud clock for the rates shown in Table II. This module also outputs digital controls and measures the gain from the acoustic computer. A parallel interface element (PIE) interfaces this hardware to the 6100 bus.*

Figure 7 shows the data decoder. This circuitry has been designed recognizing the fact that the serial data is of unknown rate and with a bit duration which can fluctuate. The data decoder can work in one of three modes, namely bit duration measurement, data duration measurement, and data conversion. During bit duration measurement the counter measures the duration of each successive bit. The microprocessor computes an average value and compares the deviation of each bit from the average value. The data duration, as shown in Figure 8, is the duration of the one and zero bits respectively. From the formation of the pulse duration modulation it will be expected that the average of the bit duration of a zero and a one bit would be close to half the average of the bit duration as measured earlier.

However, effects of the ocean path and of the filter characteristics cause all bit durations to be lengthened and therefore the point at which the value of the bit is discriminated can be computed from the average of the one and zero bits. This value is than used by a comparator to generate a bit clock pulse. This clock pulse, shown in Figure 8, is used to convert the PDM data into NRZ form. The data is read into the 6100 system for determination of its value. A simple four bit counter determines the location of each word on arrival.

It should be noted that a common method to derive the bit clock from the data is to use a phase lock loop to lock the bit clock and to generate any missing clock transitions. Due to the large amount of ocean turbulence bit jitter that is generated, it was found that a phase lock loop, with sufficient latitude to follow the bits, would no longer provide this lockover and hence the system shown in Figure 7 was devised. The system shown is extremely flexible and bit durations over an extremely wide range can easily be determined.

3.3 Acoustic Computer

In a similar manner to the data computer the acoustic computer uses a common bus structure and pre-

dominately CMOS devices to generate a low power acoustic transceiver. This computer is based on the RCA 1802 CMOS microprocessor. This microprocessor is chosen primarily because of its low power consumption and with 16 internal registors, it is possible to design a minimum component computer with PROM memory only. This PROM memory is combined with the 1802 on a single CPU board. Devices used are Intersil IM 6604 IPL CMOS PROMS and can hold 4K bits. A single 4K PROM is capable of holding far more program than is used in the particular application.

The acoustic computer has two primary functions. It interprets the control word to set constants in a synthesizer board which generates, relative to a one megahertz crystal oscillator, frequencies used in transmission and reception. This synthesizer is shown in Figure 9. Starting with a one megahertz crystal oscillator a combination of multiply and divide with two constants, J1 and J2, generate a selected frequency Fx. J1 is fixed and in the present system is equal to 128. J2 is selected from a table within the 1802 program. Eight values have been selected and these determine the basic frequency of each channel. J2 can be any integer from 1 to 165 allowing frequencies to be selected that are separated by 600 Hz (in terms of the center frequency of transmission).

3.3.1 Transmitter Operation

The actual acoustic signal transmitted is generated by a circuit shown in Figure 10. The frequency Fx from the synthesizer is combined with a serial data output to produce a square wave signal modulated in the form of the required output signal. This is formed by dividing Fx by one of two constants, N1 and N2, the choice of N1 and N2 depending on the instantaneous value of the serial data output. For the experiments performed to date N1 is eleven and N2 is thirteen so that the upper frequency is Fx/11, the lower frequency Fx/13.

To maximize the energy into the transducer it is useful to drive the transducer with a sine wave. The square wave signal generated is transformed to an instantaneous sine wave of the same frequency by the manner shown in Figure 10. A square wave generator which produces instantaneous sine and square waves of the same frequency is modulated by a control voltage generated by a phase comparator that compares the generated square wave with the referenced square wave produced earlier. Thus the sine wave modulation has zero crossing modulation and no switching transients occur. The final sine wave level is selected by a step attenuator according to the required output power and amplified by a push-pull power amplifier to drive the transducer. The maximum output power corresponds to approximately 100 volts RMS across the transducer and is constant over the range of frequencies to be transmitted.

It should be observed that the transmitted frequencies, since they are digitally derived from a 1 megahertz crystal oscillator have frequency stability to the same level of the crystal oscillator, i.e. 1 in 10^6. This allows the use of accurately controlled digital filters in the receive circuit since drift of these frequencies will not be a problem.

3.3.2 Receiver Operation

The receive signal at the transducer proceeds through the stages shown in Figure 5. The transducer output is amplified by a low noise preamplifier and routed to a pair of controlled analog filters. The output from these filters proceeds through an automatic gain control amplifier operated by the 1802 so that a constant input signal enters the set of digital filters. These isolate the upper and lower transmitted frequencies. Presence detector circuits generate a serial data input and this is routed to the data computer.

The preamplifier may be either internal to the system or in the case of the shipboard computer is mounted inside the transducer and external to the system. This preamplifier is in both cases based on a 2N 4868A FET transistor with noise equivalence of 5 NV/$\sqrt{Hz}$.

The preamplifier gain of 30 dB provides an amplified signal input to a pair of controlled analog filters (see Figure 11). The first of these is a high pass filter and is used to help reduce the larger amount of ocean background noise at the lower frequencies. The signal continues through a band pass filter centered on the center frequency of the channel with a filter Q of 8. The two filters together would produce an effective 20 dB gain of signal to noise for a flat background noise spectrum. For the decaying background noise spectrum shown in Figure 12 the signal to noise is improved approximately 30 dB. Figure 12 shows a typical combined filter response.

The signal proceeds to an automatic gain control circuit (see Figure 13). This is a programmable gain amplifier whose gain is given by the instantaneous value of the gain number GN output from the 1802. A simple program in a microprocessor detects a "too high" or "too low" level of signal and adjusts the gain accordingly. The speed of response can be adjusted by changing the delay in the program in the 1802 PROM or more simply by adjusting the microprocessor clock. Actual responses in the order of 120 dB per millisecond appear adequate. The actual gain word is six bits and the total gain control of the programmable gain amplifier is from 0 to 60 dB so that individual steps of gain of about 0.9 dB are achieved.

The constant level output from the automatic gain control circuit (0 dB $\pm$ 0.5 dB) enters a set of three digital filters as shown in Figure 14. These filters, constructed from Recticon charge coupled devices, are transverse filters whose bandwidth is fixed with respect to a center frequency. However, the center frequency is directly proportional to a clock input and by suitably selecting the clocks on the three filters it is possible to produce the filter response shown in Figure 15. It can be seen that the selectivity of the two filters is sufficiently high that over 20 dB separation can be achieved, moreover, the signal to noise over the background is greater than 30 dB. The output of these filters is fed to a special circuitry called presence detectors.

The performance of these presence detectors is that of a floating comparator (see Figure 16). Signals are generated that are proportional to the amplitude of the upper and lower frequencies in their respective filters. It is at this point that the presence of multipath signals is recognized. These multipath signals are within the narrow frequency band of the digital filters and appear as a dynamically varying noise. The presence detector circuit therefore is designed to detect the amplitude of the primary signal and the amplitude of multipath signals and to select a level between these so that the presence of the primary signal can be discriminated.

By using both the "presence" of the upper and lower frequency signals it is possible to overcome one of the basic problems concerning high Q filters. The signal entering the filter system would have a rapid on and off response, however, the filter response because of its effective Q shows a tendency to ring or retain the presence of that signal for longer than the input. This ringing can destroy the functional operation of the circuit if it is sufficient to overcome the period of absence of the signal. Measurements of the dynamic response of the filter cascade showed that the onset of the signal from the filter was extremely predictable but the decay *of the paths was considerably less predictable. Therefore the onset signals only from the presence detector circuits are used to reconstruct the serial data input to the data computer.*

It should be observed that the presence of multipath signals has one direct effect, namely that of varying the onset point of the circuit. This causes effective bit jitter of the output. This bit jitter is of the same order as bit jitter expected from ocean fluctuations. It has been incorporated into the design of the digital data decoder but does limit the upper bit rate possible.

3.4 Transducer

As discussed earlier the transducer used for both transmission and reception of signals was constructed in house. Cylindrical piezo ceramic elements were used. Four elements, each 1½ inches diameter and ½ inch length were assembled into a single cylindrical form and connected in series with alternate elements of reversed polarity. This transducer is capable of transmitting up to 10 watts output power. Resonant frequency is about 31 kHz. Because the transducer is used to transmit, no internal preamp is used in the transducer. Figure 17 shows a picture of the underwater housing of the submarine system with one of these transducers mounted on the top surface. For the ship system the same transducer was used at the end of some 50 feet of coax cable suspended over the vessel used for the experiments.

While it was not possible in the present time scale to accurately calibrate the transducer for both transmit and receive characteristics a short calibration experiment was run where two transducers were suspended a known distance apart and a short duration signal transmitted from one to the second. From these it is possible to derive a transfer function, i.e. the product of the transmission response and the receive response at one meter. This transfer function has a response of -57 dB at 27 kHz. The transfer function was reasonably flat from 10 to 100 kHz. If the receive response to the transducer is calculated then a value of 186 dB re 1 V @ 1 μPa is attained and this leads to a value of 129 dB re 1 μPa @ 1 V for the transmitter. While these values are not exact they do allow the evaluation of the observed signal and noise to see whether they are the expected values.

As seen in Figure 19, the submarine package is a short cylinder with approximately 1,000 psi pressure capability. Figure 18 shows the internal computer construction. The various modules are an 4½ x 6½ inch cards packaged onto a bus in conventional manner. The packaging is capable of 200 G shock and can withstand the normal rough treatment seen on board ship. The package is almost neutrally bouyant and can be easily deployed as part of a short suspended array off the bottom or suspended below a drifting buoy on the surface.

4. SOFTWARE DEVELOPMENT

4.1 Methodology

The development of microprocessor based systems requires the simultaneous construction of hardware and software. The debugging of any system requires a method of isolating problems concerning one or the other. The 6100 microprocessor, as discussed earlier, responds directly to the instruction set of the PDP 8 computer. To expedite this software development a PDP 8E is used and a direct connection between this PDP 8 and the microcomputer memory has been constructed. This allows the PDP 8 to transfer a program, after checkout, directly into the memory of the microcomputer. Modified programs, with break points, can also easily be used to identify hardware problems in the microcomputer. The effective speed of software development resulting from this helps enormously. Figure 19 illustrates this connection. Programs can be written, are stored into the disk memory of the PDP 8, tested in the microcomputer and then the resulting program permanently placed into PROM using a PROM programmer directly built into the PDP 8. As an example of the speed of software development a 1K program has been written, debugged and burned into PROM in less than a four hour period.

The 1802 does not use an instruction set which can operate on the PDP 8. A cross assembler was used, however, the resulting program was wasteful of memory space. Fortunately the 1802 programs are sufficiently short that they can be written directly into PROM.

Although high order programming is readily available the kind of data manipulation currently used in this system is sufficiently simple that programs are written directly in assembly language. This results in a highly compressed program and very careful attention can be paid to software timing. In many instances this has to be consistent with the data timing.

4.2 Acoustic Computer Program

As discussed earlier the acoustic computer, using an 1802, has a very simple program. There are two main subsections; on its initiation, commanded from the data computer, the acoustic computer will read the control word and generate, using an internal table, the correct quantities for the synthesizer. It will then proceed, during receiver operation to operate the automatic gain control. The gain control is shown in Figure 20. Extreme simplicity of this program is necessary since it is desired to have very rapid response.

this program is in actual fact in use elsewhere for data collection both from multiple parallel sources and from a large number of sensors interrogated on one serial conductor.

The software associated with the data transmission has been developed. The exact nature of this depends on whether the software is associated with the shipboard acquirer of data or of a submarine transmitter of data. The software elements, however, are common to both. The exact procedure and pattern used will be different.

The receive transmission process for the submarine package will involve listening for data by scanning the channels to determine if a signal is present, examining the various processes to determine if the channel is adequate followed by adaption of the channel to new parameter values and the subsequent transmission of data. The adaptability process whereby the channel characteristics are analyzed and a new channel discerned are the result of this coming year's activity.

Figure 21 shows a block diagram of this software. During initialization the scratch pad memory is *initialized with constants and subroutine entries. In the case of the shipboard package the switch register is examined to determine the initial transmission. If a transmission is required than the process of constructing the calibration frame takes place. According to the channel and bit rate the 1 second bit clock is constructed and this and the calibration frame transmitted. The program then proceeds to a "prescan".*

This "prescan" scans the channels, measures the gain of the signal and then stores it in memory as a measurement of the signal. It is assumed at this point that no signal is incoming and that this gain corresponds to the background noise. During the scan each channel is again sampled and the gain of each channel determined. Analysis of the signals determines whether any of them are above the noise level by a required margin.

If so, then the channel is selected and successive incoming bits are examined to determine whether four successive bits have a bit duration whose scatter is within tolerance. This bit tolerance has been selected during early tests to be 1/8 of the bit duration. If bit coherence does not exist after three attempts then it is assumed that an outside noise source has been detected and the new background noise level recorded.

If bit coherence is detected then the program proceeds to a data duration measurement. Here the duration of the one and the zero bit is determined and this allows the bit duration comparator to be set so as to convert from PDM to NRZ. The data transmission is initialized by a sync word (a known twelve bit pattern) and the program proceeds to examine whether it can receive the sync word within a certain number of bit counts. If the sync word is received then the word count is set to zero and data is read. If the sync word is not detected than the program will proceed to the adaptability process.

During the tests of some of the hardware and software the submarine package was designed to be periodically initialized and hence transmit. The receiver operated a modified version of the software and the data associated with the channel was recorded.

The adaptability procedure has, from the existing receiver software, two entry points depending on whether or not the sync word is received. If the sync word is not received then the receiver does know that a message was attempted and it knows certain aspects of the received signal. If the sync word is received, then *in addition to the above parameters the transmitted power level, the message status and the bit error rate can also be determined. The receiver is then one step further along in the adaptability process. This information, the entry point to the adaptability procedure is outlined in Table IV. It should be emphasized that the determination of a coherent bit duration is an extremely important factor in the determination and evaluation of the acoustic data channel.*

5. EXPERIMENT RESULTS

The system described so far represents a reasonably high degree of sophistication. To build such a system takes a period of time during which various aspects of the system are evaluated and checked. The experiments performed to date have aimed primarily at evaluating the various subelements of the system, to determine their adequacy. These tests can be subsectioned into various components, they are:

(1) lab tests of the digital electronics;
(2) lab tests of the transmitter operation;
(3) lab tests of the receiver operation;
(4) combined tests in the lab;
(5) sea operational tests to determine various parameters.

5.1 Laboratory Tests

The digital electronics described was operated in the lab by allowing the transmitter digital section to input data directly into the receiver digital section. Here the software and the hardware were evaluated to determine that data would be received and evaluated. By direct hookup the influence of any noise present is removed and it was possible to quickly determine that the digital electronics worked to specifications. Two basic conclusions were reached. First the maximum bit rate that could be achieved was dependent on the timing rate in the sync word loop. This loop, as originally designed, took 148 microseconds to perform allowing for a maximum bit rate of approximately 6700 kHz. This period could be improved by modifying the clock used to drive the microprocessor. For systems currently imployed this is 1 megahertz. A higher frequency clock would allow us to easily reduce the timing loop to less than 100 microseconds, sufficient for 10 kilobaud.

An additional problem emerged mainly that if the sync word was not received by the end of the data stream the system would remain hung up waiting for it. Adding a bit counter was not a full proof method of solving this since an incomplete transmission or signal loss could result in insufficient data bits being transmitted.

For the present system where data was merely transmitted from a submarine package to a shipboard package the hang up of the receiver in a sync wait loop was not considered detrimental since manual override could occur. The design modification for an external kickout clock will be included in future work.

The transmitter operation was evaluated in the laboratory. The modulator shown in Figure 10 was found to work perfectly and a power amplifier capable of 100 V RMS output was designed. Over the wide frequency range shown, problems initially occured with parasitic oscillations due to stray feedback, however, these were suppressed without loss of frequency response.

The receiver was evaluated in the laboratory. The preamplifier was calibrated. 120 dB was the effective maximum gain of the total system and therefore input noise of less than 1 microvolt is desired. Initially noise levels of three to four microvolts were observed and this was found due to noise pickup from the digital electronic circuitry. Some redesign of the system to isolate this has effectively reduced the noise level to below 1 microvolt in all channels.

The analog filters were individually calibrated and adjusted and tests on the automatic gain control with dynamically varying input signals showed an adequate response with no signs of parasitic oscillations. The digital filters were calibrated, an example of the calibration is shown in Figure 15.

The transient operation of the digital filters was also carefully measured. This transient response is the key to the maximum baud rate that can be achieved with this discriminator. Figure 22 shows the response to pulsed sinusoidal wave forms. To establish a criteria whereby this transient response provides a limit to the maximum data rate is somewhat difficult. If the half amplitude point of the response is taken as the criteria for the limit then a miximum bit rate of 600 baud at 10 kHz acoustic frequency and 1.2 Kbaud at 90 kHz acoustic frequency is reached. In actual practice the discriminator works to higher frequencies than this. Table VI shows the maximum bit rates that were achieved in the laboratory with the transmitter output suitably attenuated and fed directly into the digital filter. It can be seen that 3 Kbaud can be achieved at the upper channel with suitable reduction as the acoustic frequency declines.

A final system test was made in the laboratory by actually transmitting acoustic data across the room and observing the effective response for the system. The bit rates of Table VI were achieved so that the digital filter does remain the bit rate determining factor.

Some sea experiments have been made with the hardware. These tests were made in the coastal waters off Miami, Florida in water depth of 30 to 60 meters and with maximum ranges of 2 kilometers. These tests were meant to be preliminary in nature to determine the system function and to evaluate signal and background noise levels. (These tests were mostly performed with an earlier version of the system with different assigned channel frequencies, the lower limit being 8 kHz and the upper one being 120 kHz.) An example of a recorded AGC signal in analogue form is shown in Figure 23. The transmitter had been programmed to transmit a sequence of data at three bit rates and at four output powers. This sequence is clearly seen in the data, the automatic gain circuit clearly works. Recordings were made of the digital signal from the decoder.

It should be observed that these results do show one key feature. With the present system the background noise levels were exceeding high and were some 30 dB above those summarized by Urick[5] (see Figure 24). The range at which data could be satisfactorily received was limited. It was decided that this noise source originated in the hydrophone assembly and could be eliminated with suitable rework of the system. Therefore continued *testing was postponed until this noise source could be eradicated. Actual signals received were within 5 dB of those predicted from a simple spreading absorption calculation and therefore considered quite workable.*

The criteria for establishing adaptive reasoning within the computer has been started with the basic two starting points for this adaptive reasoning is well defined. It is expected that by using these starting points, a fully adaptive system can be generated.

The final system might seem somewhat complex. This is necessary because of the level of intelligence of the system. It is well within the level of complexity of modern microcomputer assemblies and perhaps is indicative of the trend of oceanographic instrumentation.

For a system to be operational certain additional factors have to be incorporated to prevent the system from being caught in an unresolved decision loop. This additional hardware complication is not felt to be overly critical. During the coming year an effort will be made to generate a video picture and transmit this using a system based on the hardware described in this report.

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1. "An Experiment in High Rate Underwater Telemetry"; Chauncey S. Miller and Carl E. Bohman, Speery Rand Research Center.

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3. "Swept Carrier Acoustic Underwater Communications"; A. Zielinski and L. Barbour, Memorial University of Newfoundland.

4. "Air Deployed Oceanographic Mooring (ADOM) Progress Report for 1978"; Dr. Eric J. Softley (and others) (April 1979)

5. "Principles of Underwater Sound", Robert J. Urick McGraw-Hill Book Company.

TABLE I

	Typical Horizontal Range	Water Depth	Data Quantity (bits) and Data Rate (baud)	Lifetime (Battery Capability)
Non-Destructive Testing	0 - 100 M	30 - 300 M	10^5 @ 100	1 Year
Unmanned Vehicle	0 - 5 KM	30 - 300 M	Control 10^3 @ 100 Monitor 10^4 @ 100 Video @ 20 K	1 Day
Air Deployed Interrogation	0 - 2 KM	30 - 300 M	10^5 @ 2 K	6 Months - 1 Year

KEY SCENARIO CHARACTERISTICS

TABLE II

SOME THEORETICAL LIMITS

CHANNEL	CENTER FREQUENCY (kHz)	MAXIMUM BIT RATE (baud)	SLANT RANGE (Signal 6 dB over Shallow Water Noise)
0	13.7	1050	12 KM
1	18.2	1400	10.8 KM
2	23.4	1800	8 KM
3	30.6	2350	6 KM
4	40.4	3100	4.5 KM
5	52.1	4000	3.4 KM
6	67.7	5208	2.6 KM
7	87.9	6761	2.3 KM

TABLE III

RN		0	1	2	3	4	5	6	7
CN	FX	BR--------							
0	164	151.9	207.2	284.9	379.9	455.8	569.8	759.7	1139.6
1	219	202.6	276.3	379.9	506.5	607.8	759.7	1013.0	1519.4
2	281	260.5	355.2	488.4	651.2	781.4	976.7	1302.3	1953.5
3	367	340.0	463.6	637.5	850.0	1020.0	1275.0	1700.0	2550.0
4	484	448.5	611.6	841.0	1121.3	1345.6	1681.9	2242.6	3363.9
5	625	578.7	789.1	1085.1	1446.8	1736.1	2170.1	2893.5	4340.3
6	813	752.3	1025.9	1410.6	1880.8	2256.9	2821.2	3761.6	5642.4
7	1055	976.6	1331.7	1831.1	2441.4	2929.7	3662.2	4882.9	7324.3
RD		15.0	11.0	8.0	6.0	5.0	4.0	3.0	2.0

BIT RATES FOR EACH CHANNEL (CN)

TABLE IV

CHANNEL WORD DESCRIPTION

MSB LSB

D_0	D_1	D_2	D_3	D_4	D_5	D_6	D_7	D_8	D_9	D_{10}	D_{11}
CN			M	PN		MS		RD			

CN - 3 bits giving channel number (See Table II)

M - 1 bit giving coding

0 = PDM
1 = SFM

PN - 2 bits giving power level

3 = FULL POWER
2 = " " - 6 dB
1 = " " - 12 dB
0 = " " - 18 dB

MS - Message Status

0 - Channel OK, Data Sent
1 - Channel Unknown, Request Data
2 - " "
3 - Channel OK, Request Data

RD - Baud Rate Divider (See Table III)

TABLE V

INPUT INFORMATION FOR ADAPTABILITY PROCESS

FACTOR	BIT COHERENCE DETERMINED NO SYNC WORD	BIT COHERENCE and SYNC WORD
Message Attempted	X	X
Channel Used	X	X
Received Signal Level	X	X
Background Level	X	X
Bit Rate	X	X
Bit Jitter	X	X
Transmitted Power		X
Message Status		X
Bit Error Rates		X

TABLE VI

EXPERIMENTALLY DETERMINED MAXIMUM RATES OF TRANSVERSE FILTER DISCRIMINATOR

CHANNEL	CENTER FREQUENCY (kHz)	BIT RATE (baud)
0	13.7	570
1	18.2	760
2	23.4	977
3	30.6	1275
4	40.4	1682
5	52.1	2170
6	67.7	2821
7	87.9	2930

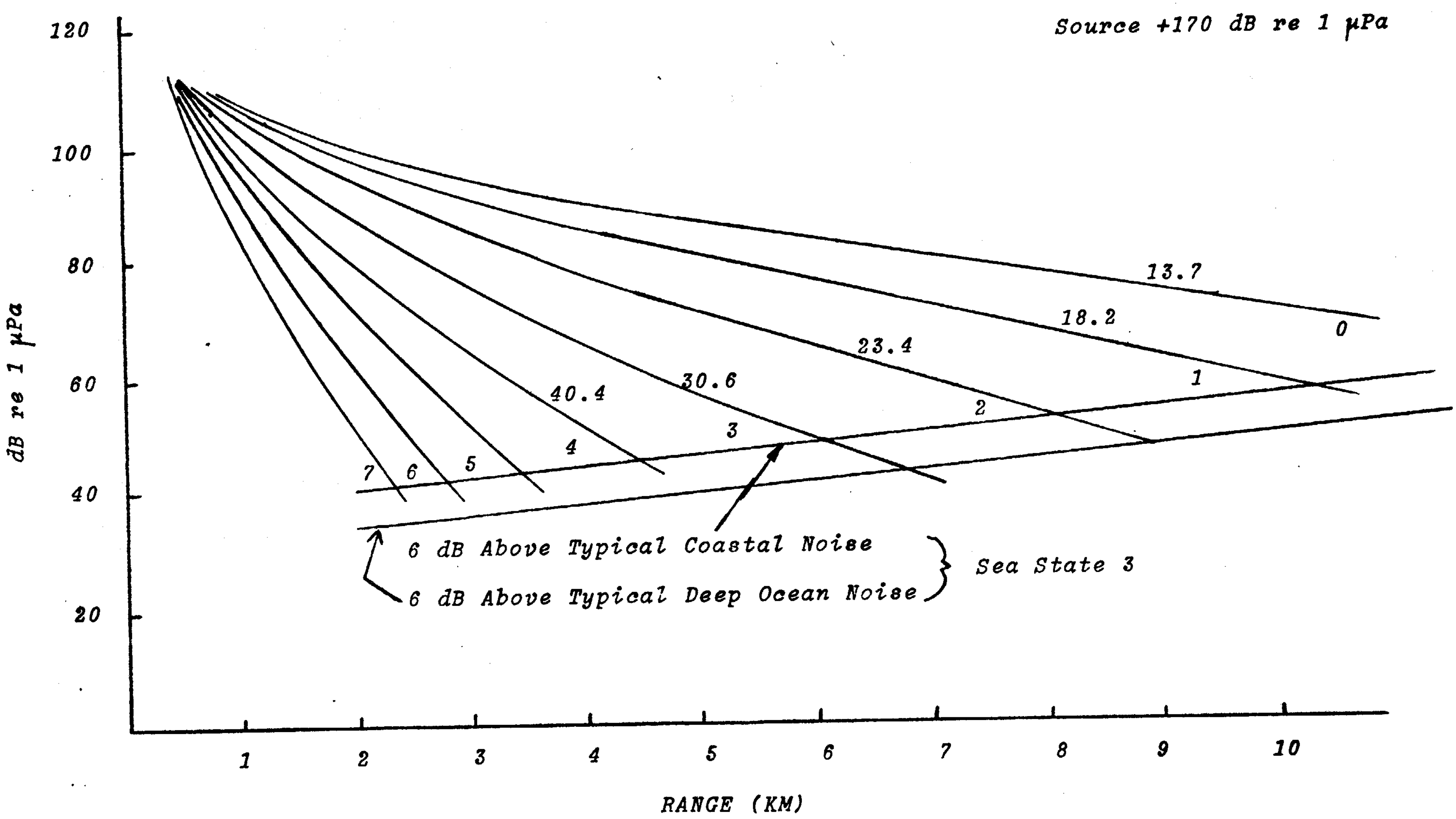


Figure 1 - EFFECT OF SPREADING AND ABSORPTION LOSS FOR CHANNELS 0 - 7

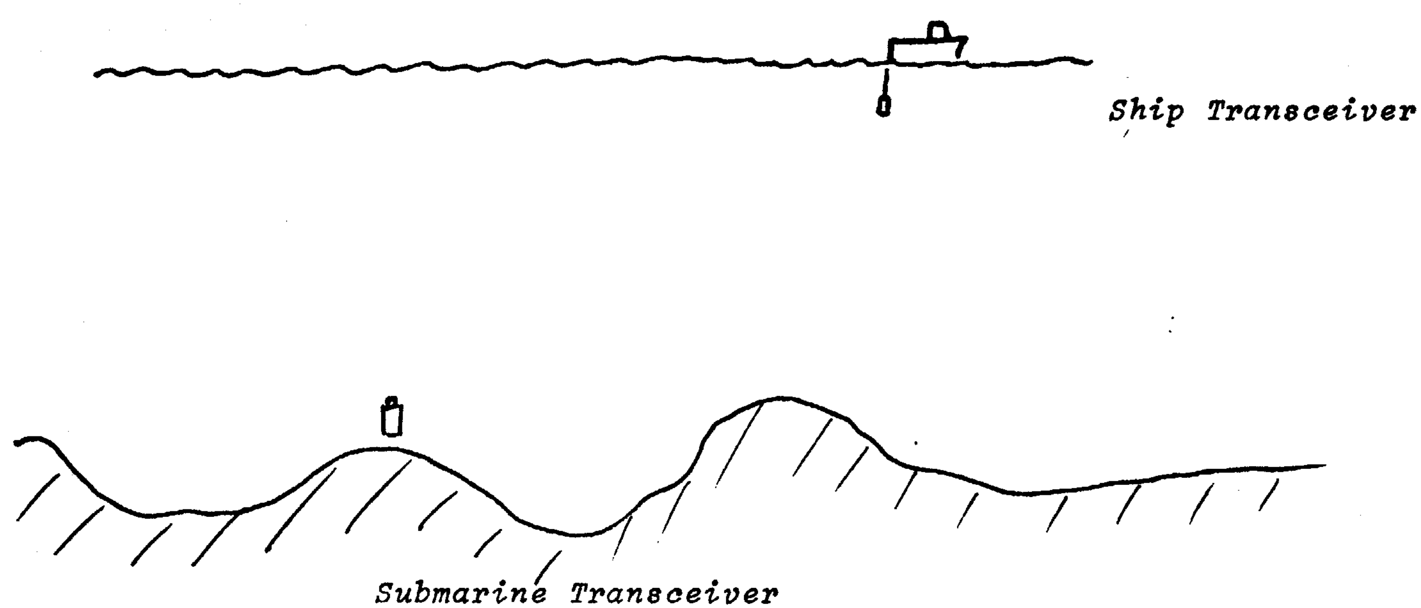


Figure 2 - TYPICAL DATA TRANSFER SCENARIO

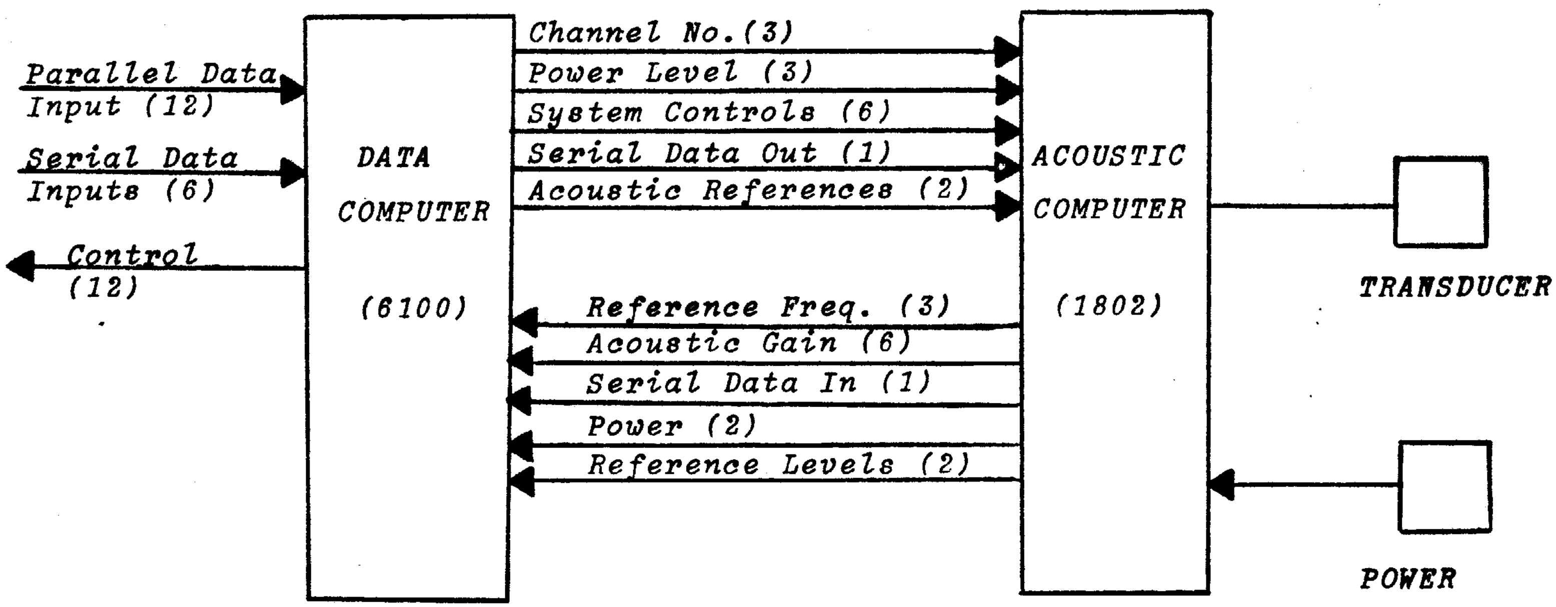


Figure 3 - SYSTEM LAYOUT (EACH TRANSCEIVER)

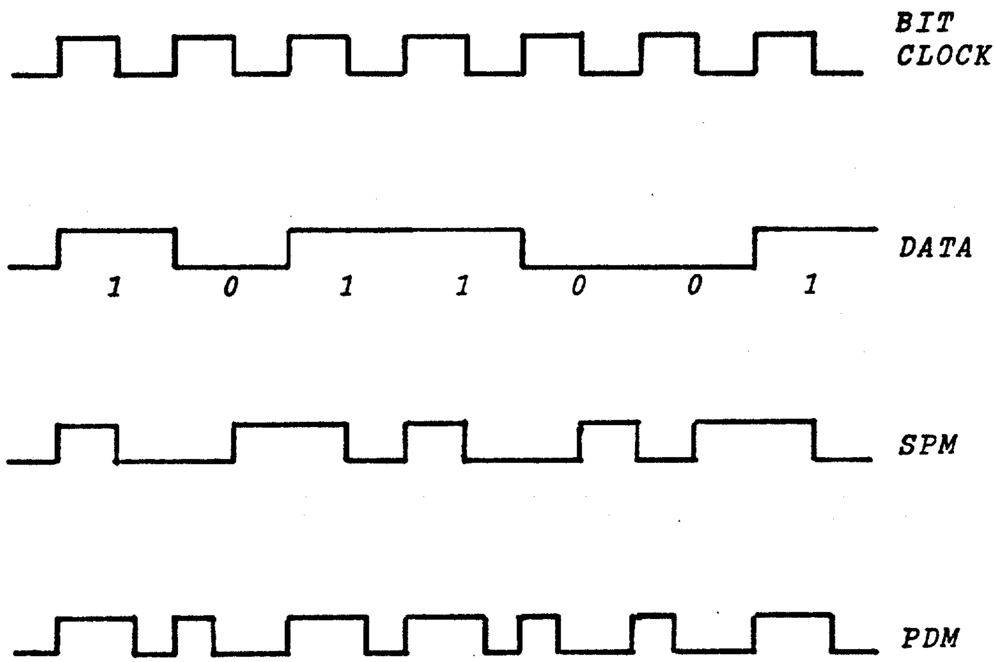


Figure 4 - SPLIT PHASE MODULATION (SPM) AND PULSE DURATION MODULATION (PDM)

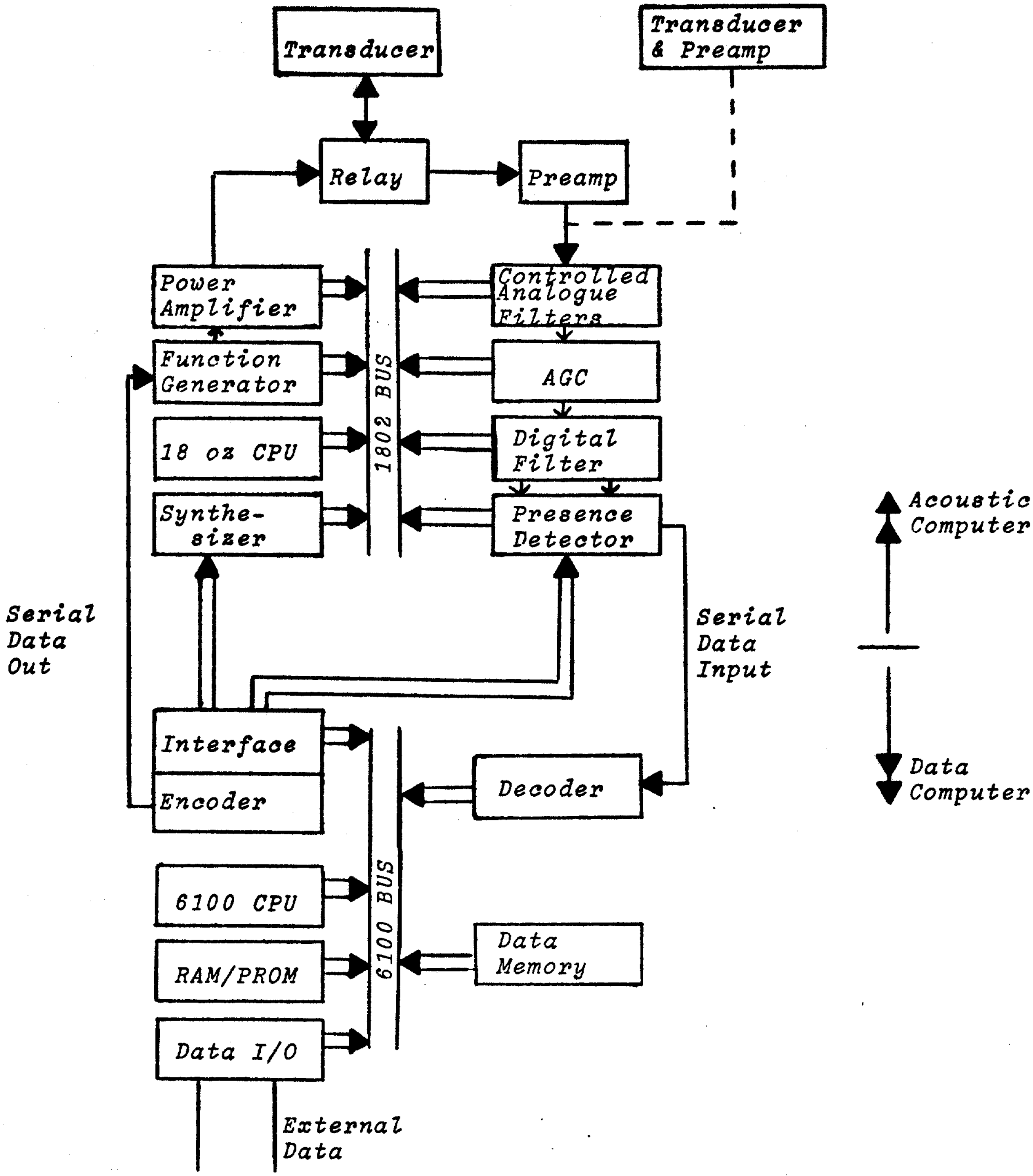


Figure 5 - TRANSCEIVER BLOCK DIAGRAM

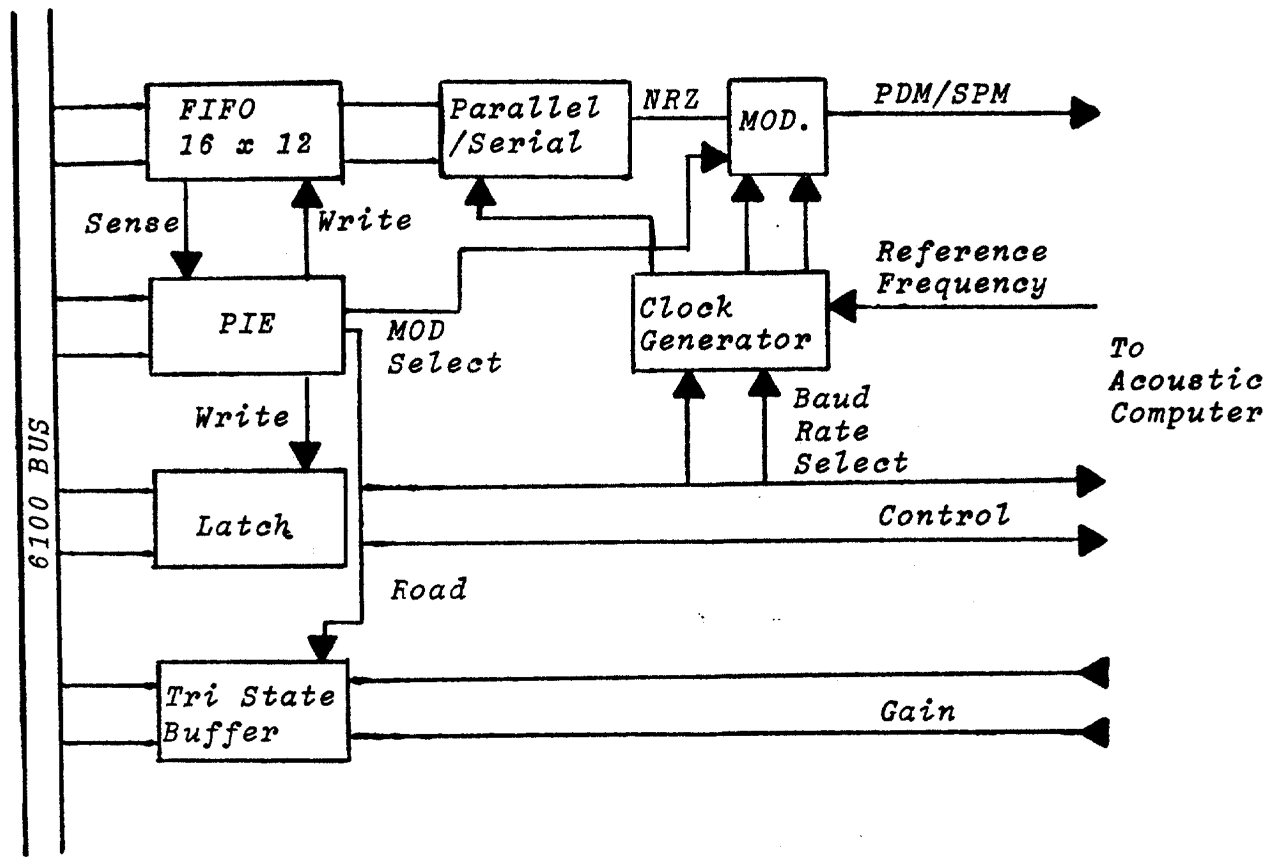


Figure 6 - DATA ENCODER

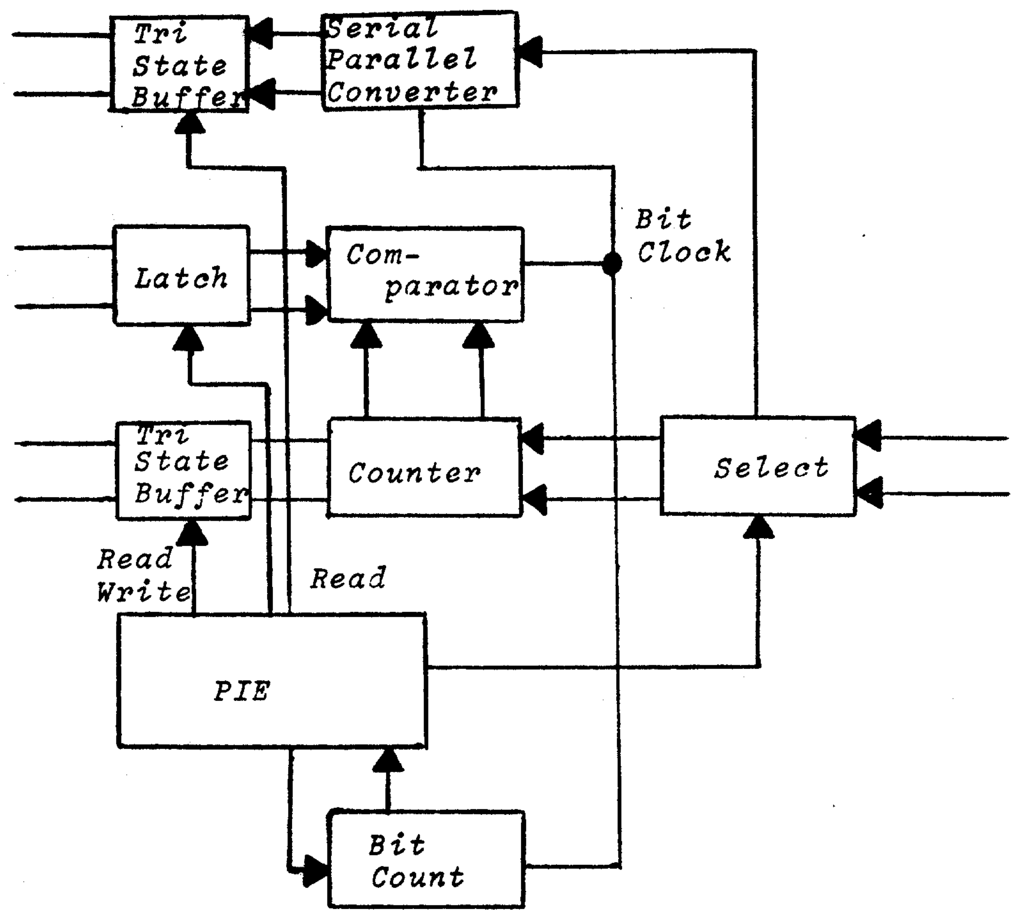


Figure 7 - DATA DECODER

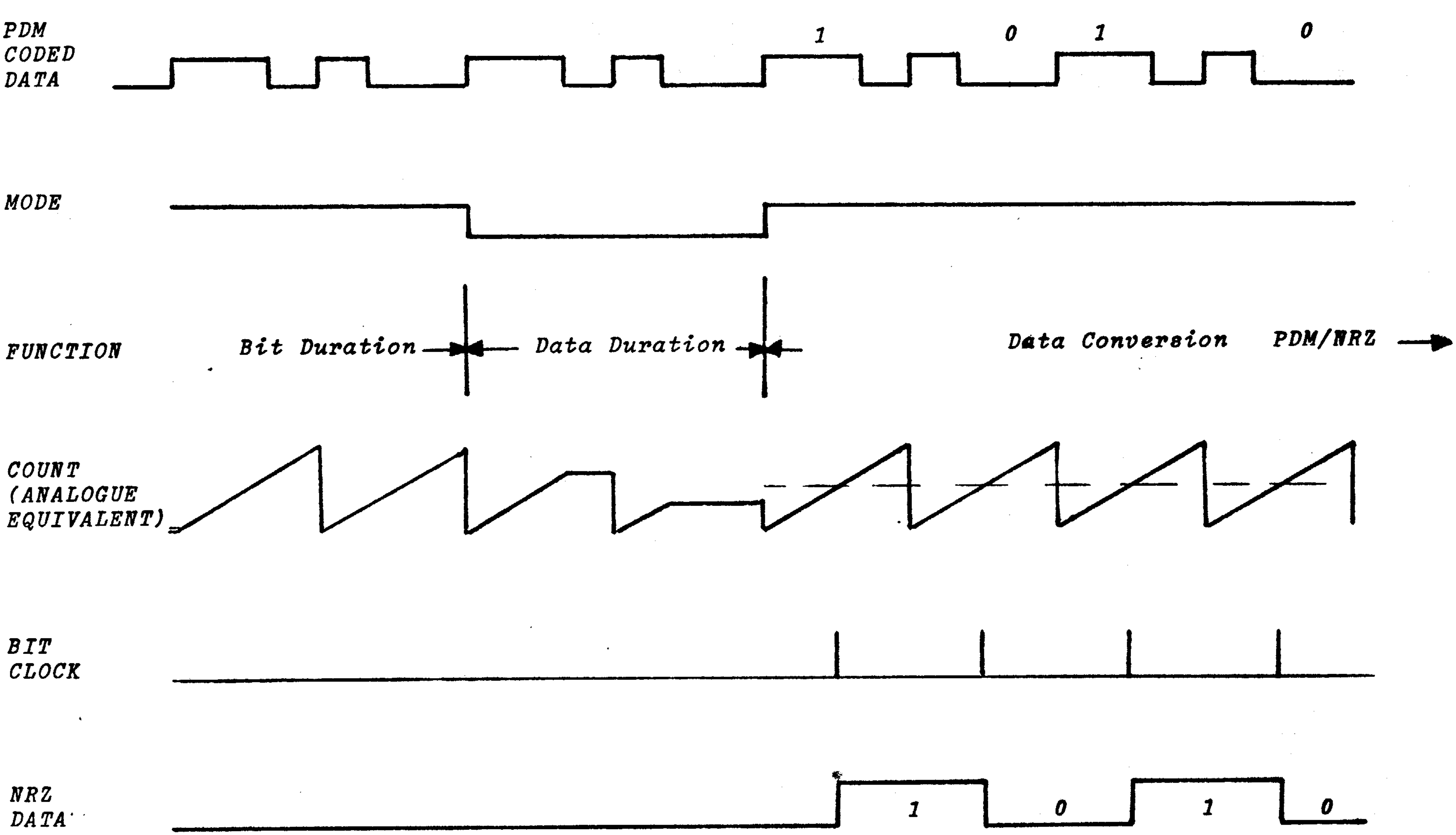
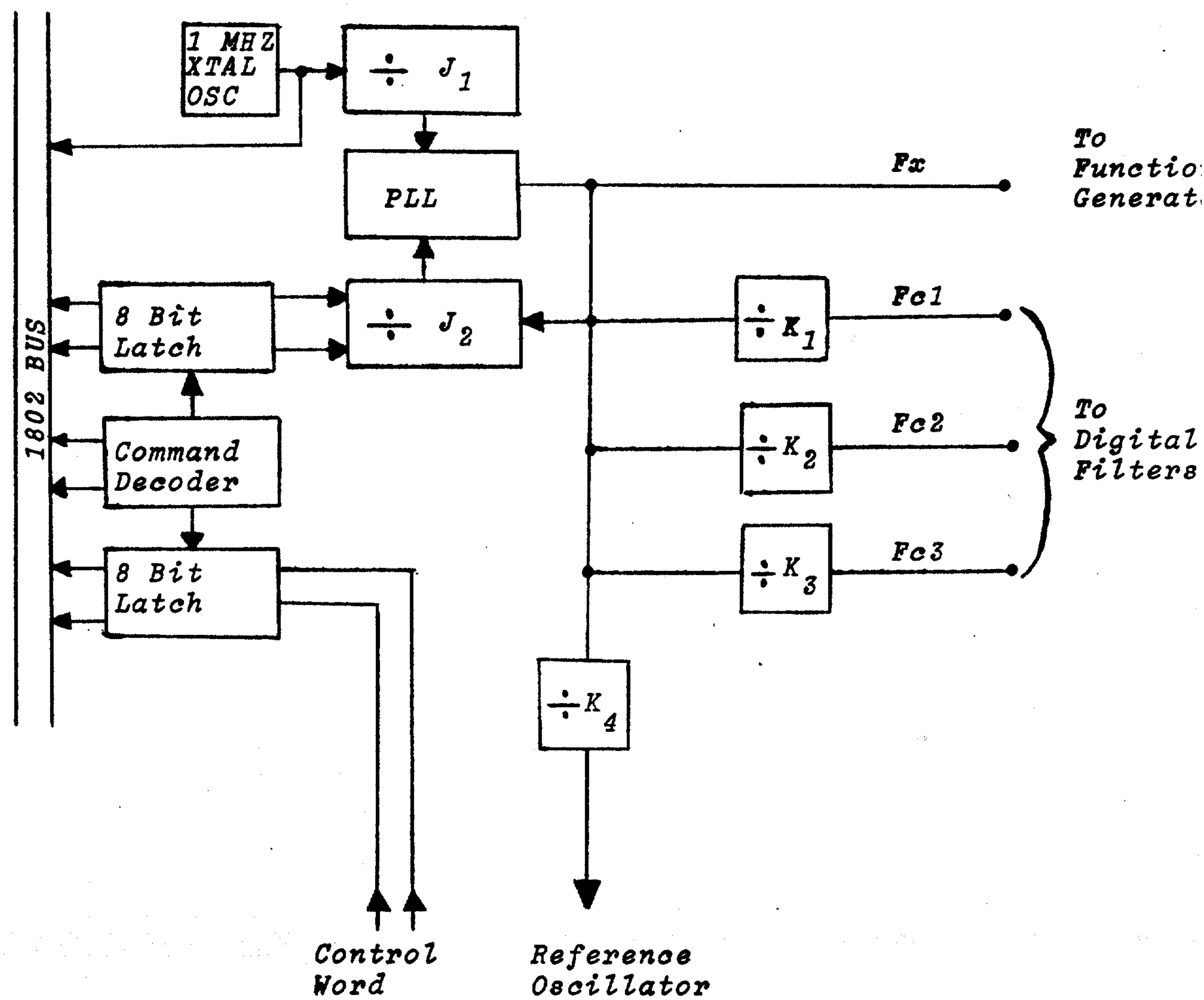


Figure 8 - DATA DECODER OPERATION



Connections to Data Computer

Figure 9 - SYNTHESIZER

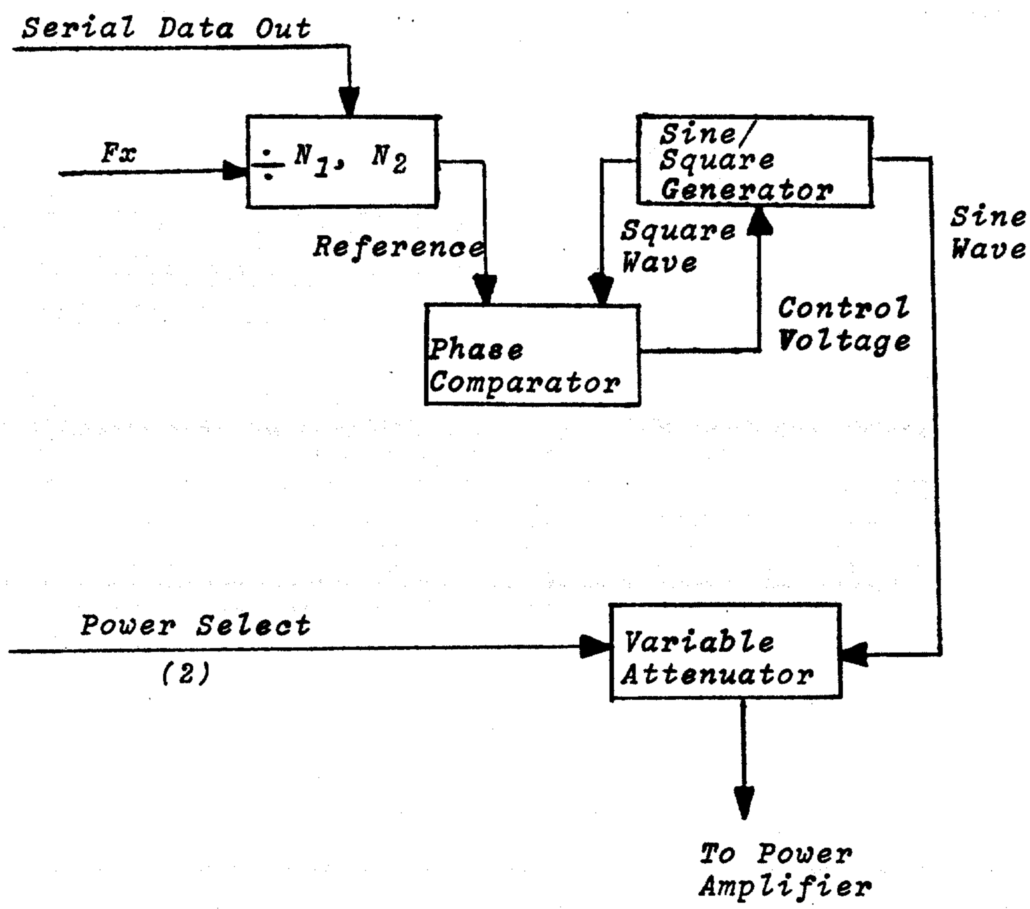


Figure 10 - TRANSMISSION SIGNAL GENERATOR

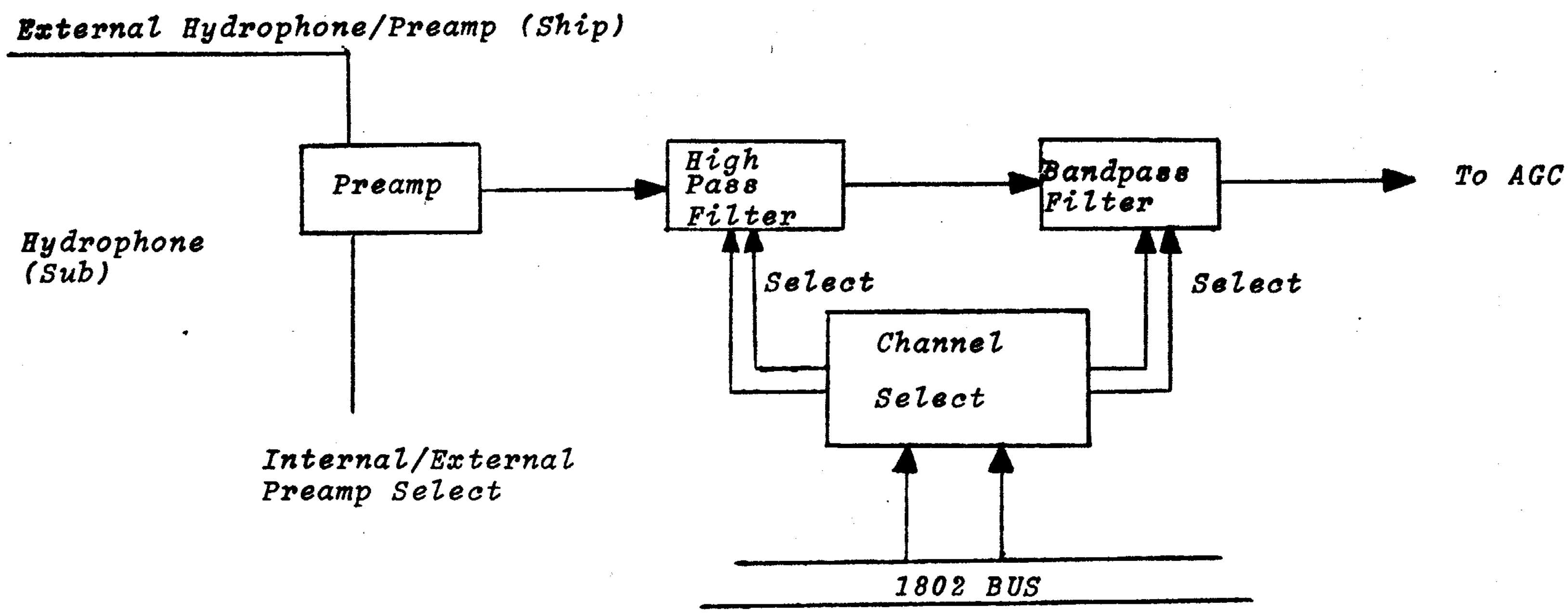


Figure 11 - PREAMP - FILTER ARRANGEMENT

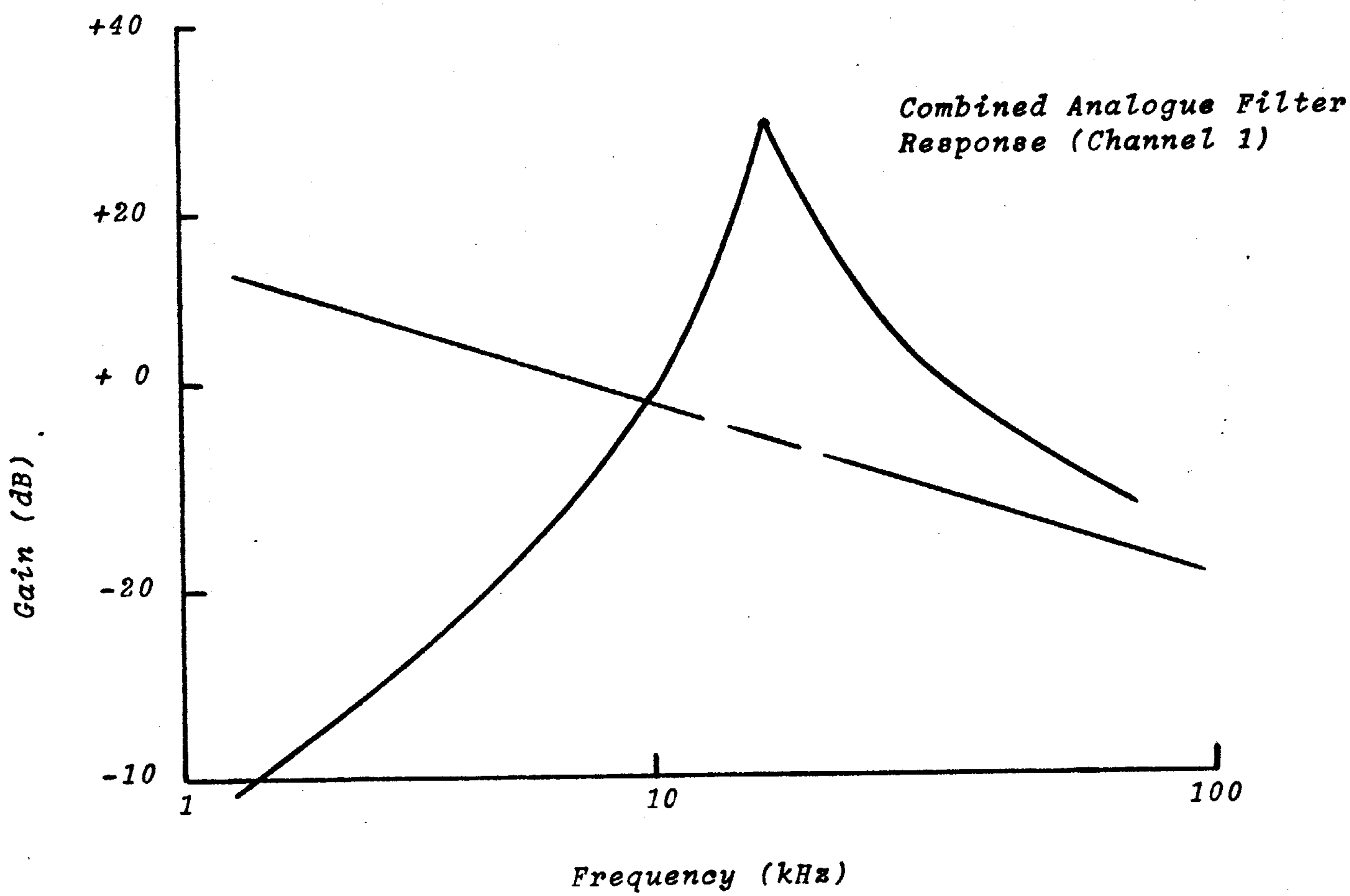


Figure 12 - TYPICAL ANALOGUE FILTER RESPONSE

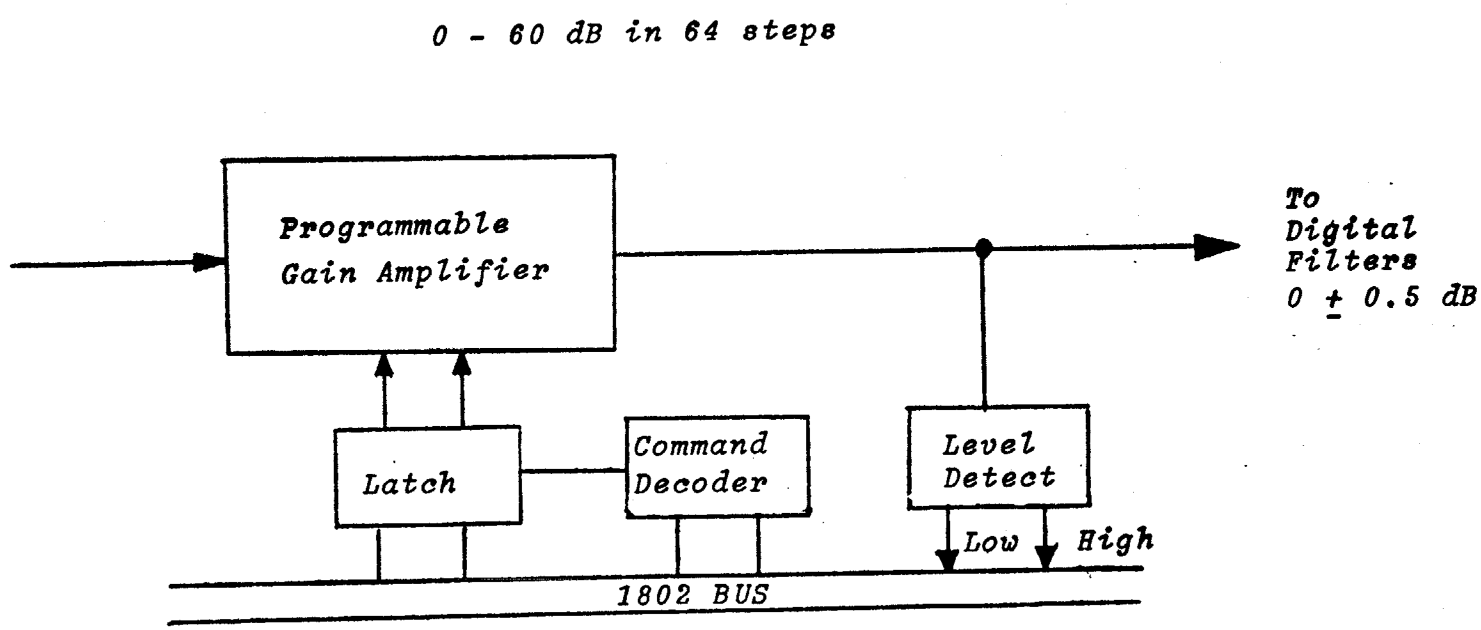


Figure 13 - AGC

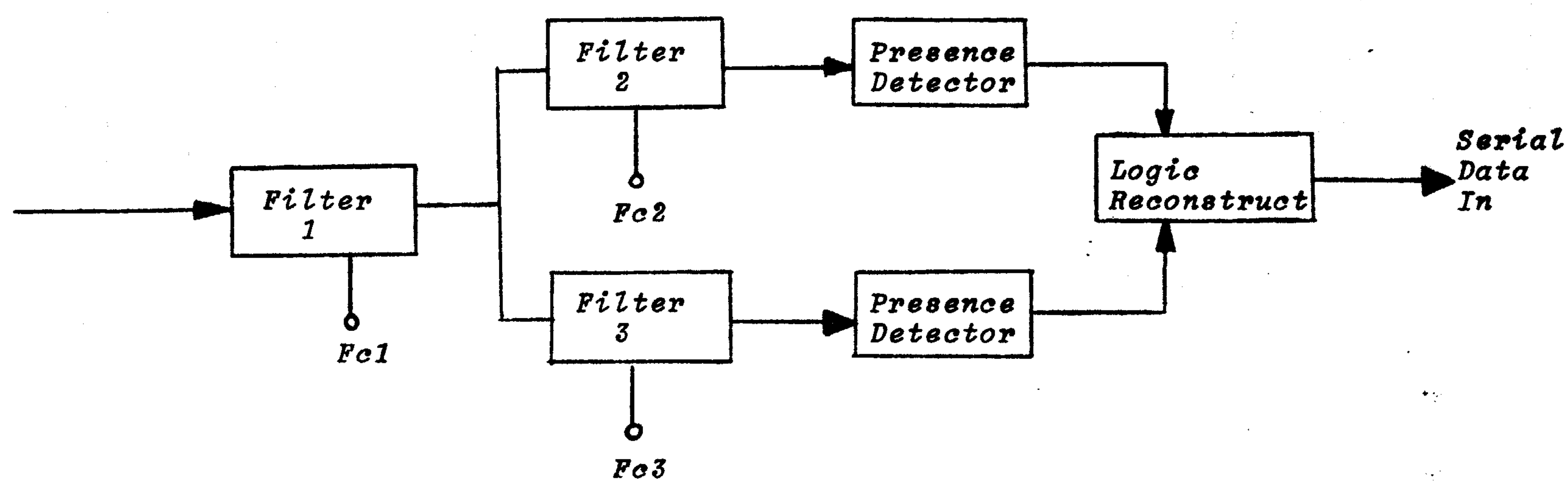


Figure 14 - DIGITAL FILTER ARRANGEMENT

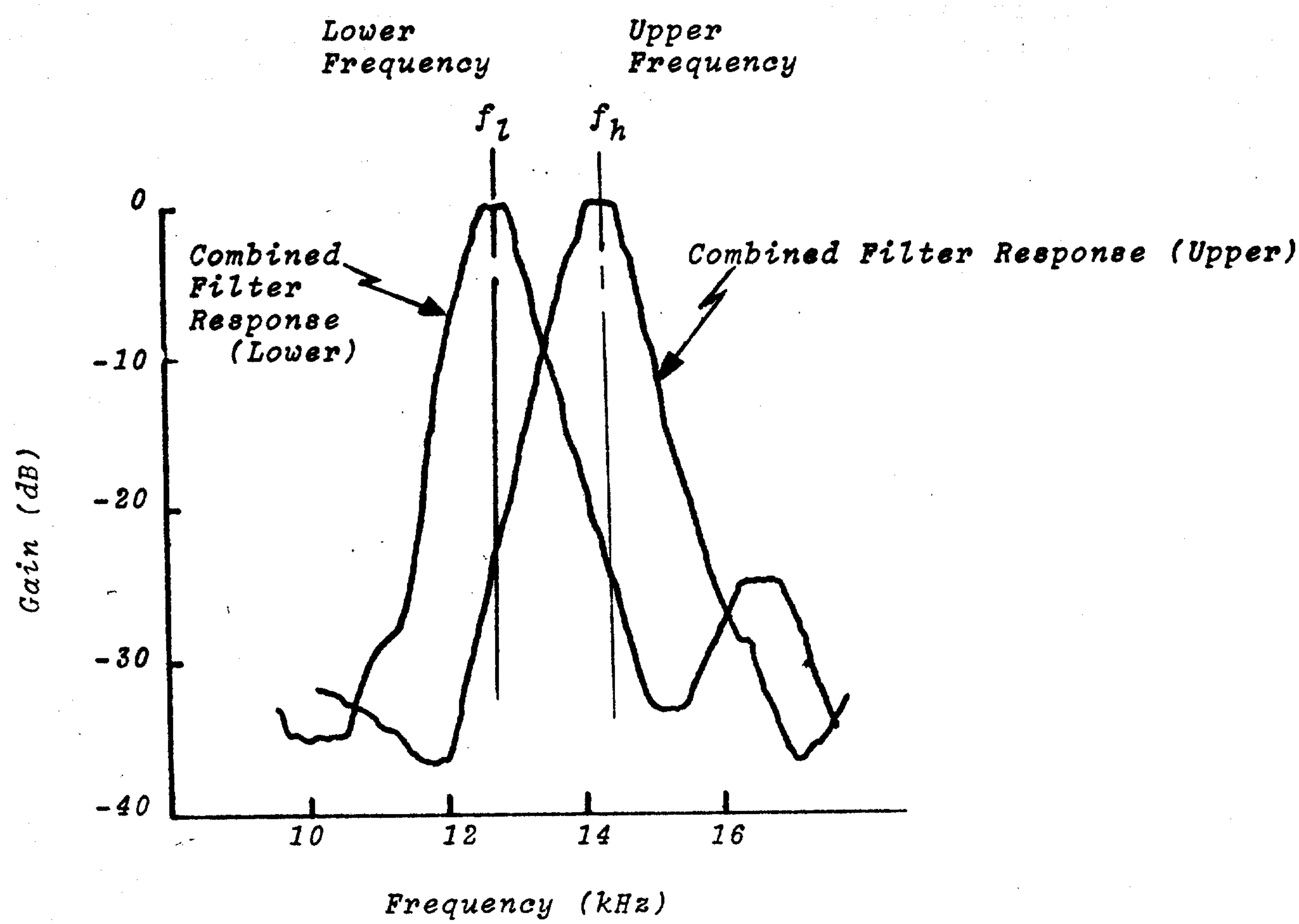


Figure 15 - EXAMPLE OF MEASURED FILTER RESPONSE FOR CASCADED FILTER

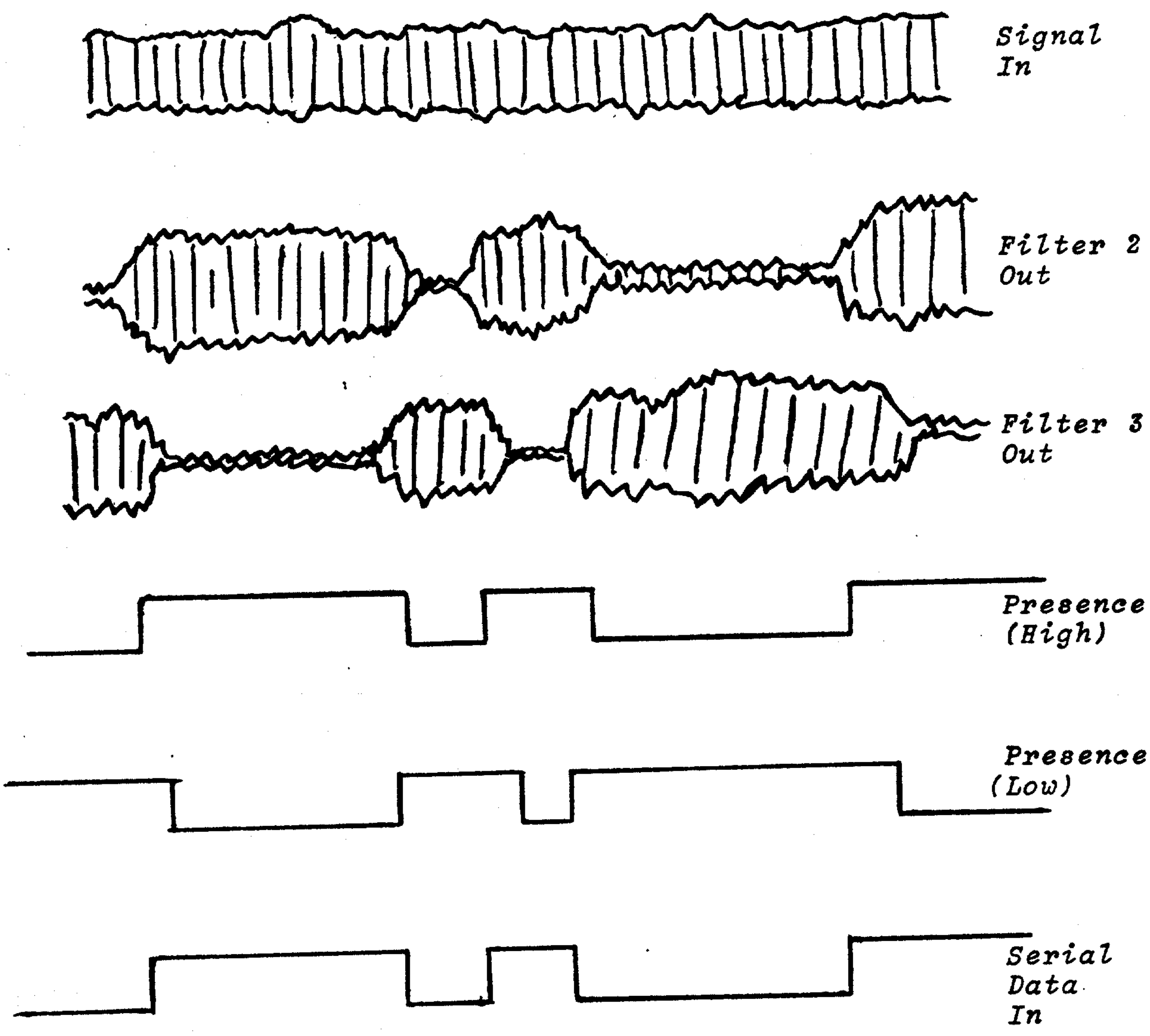


Figure 16 - LOGIC RECONSTRUCTION

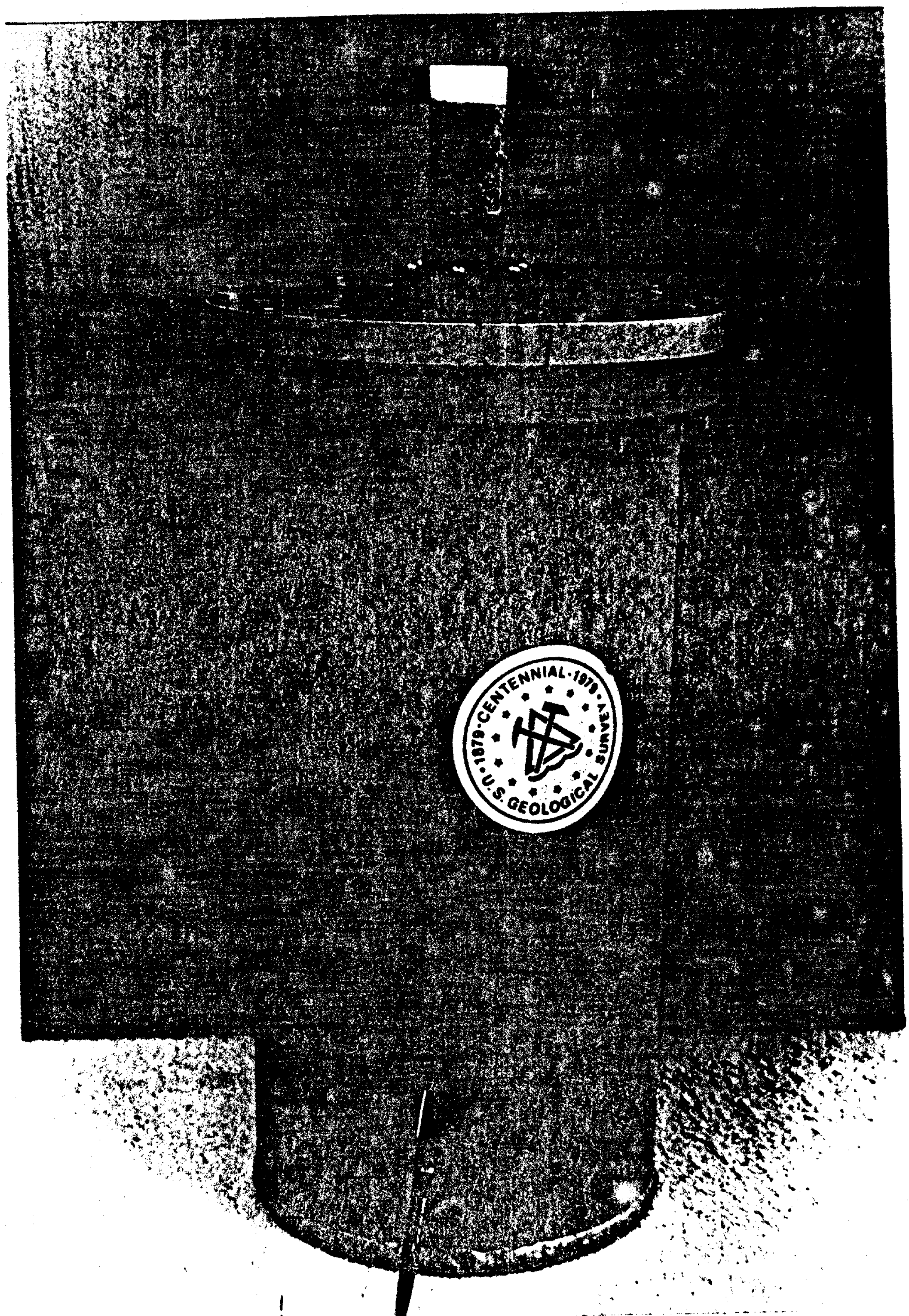


Figure 17 - SUBMARINE TRANSCEIVER

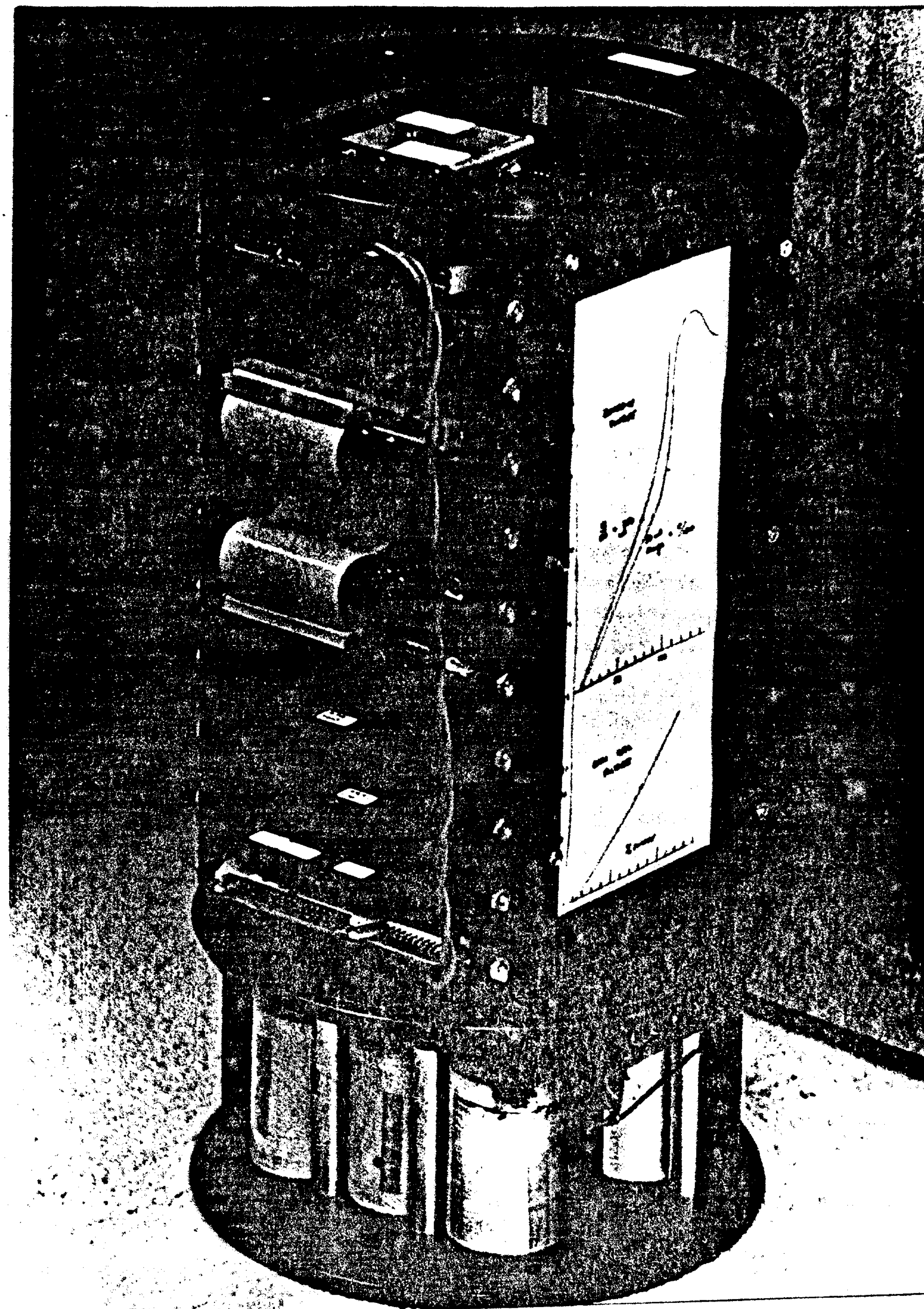


Figure 18 - INTERNAL ELECTRICAL ASSEMBLY

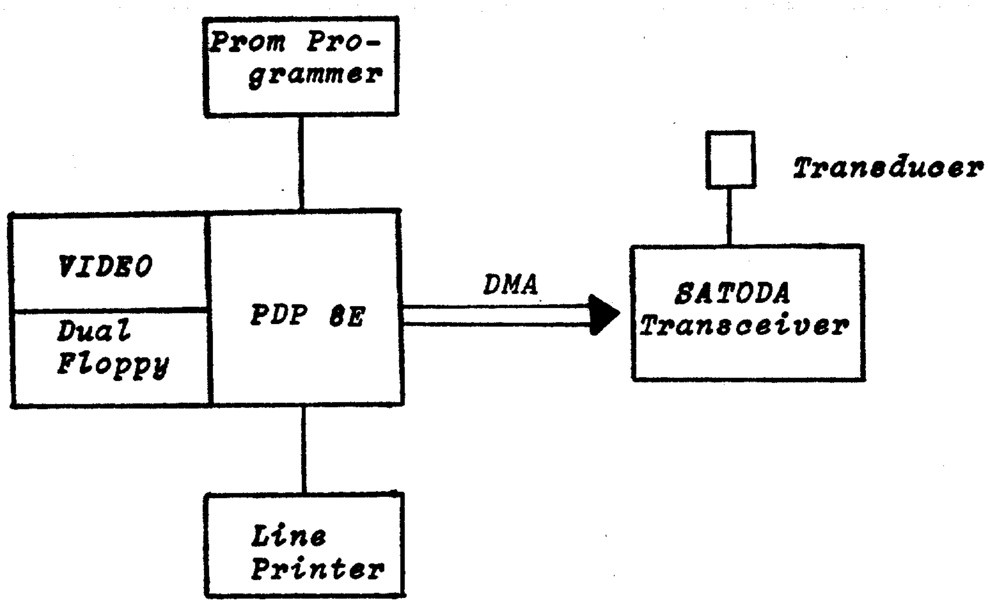
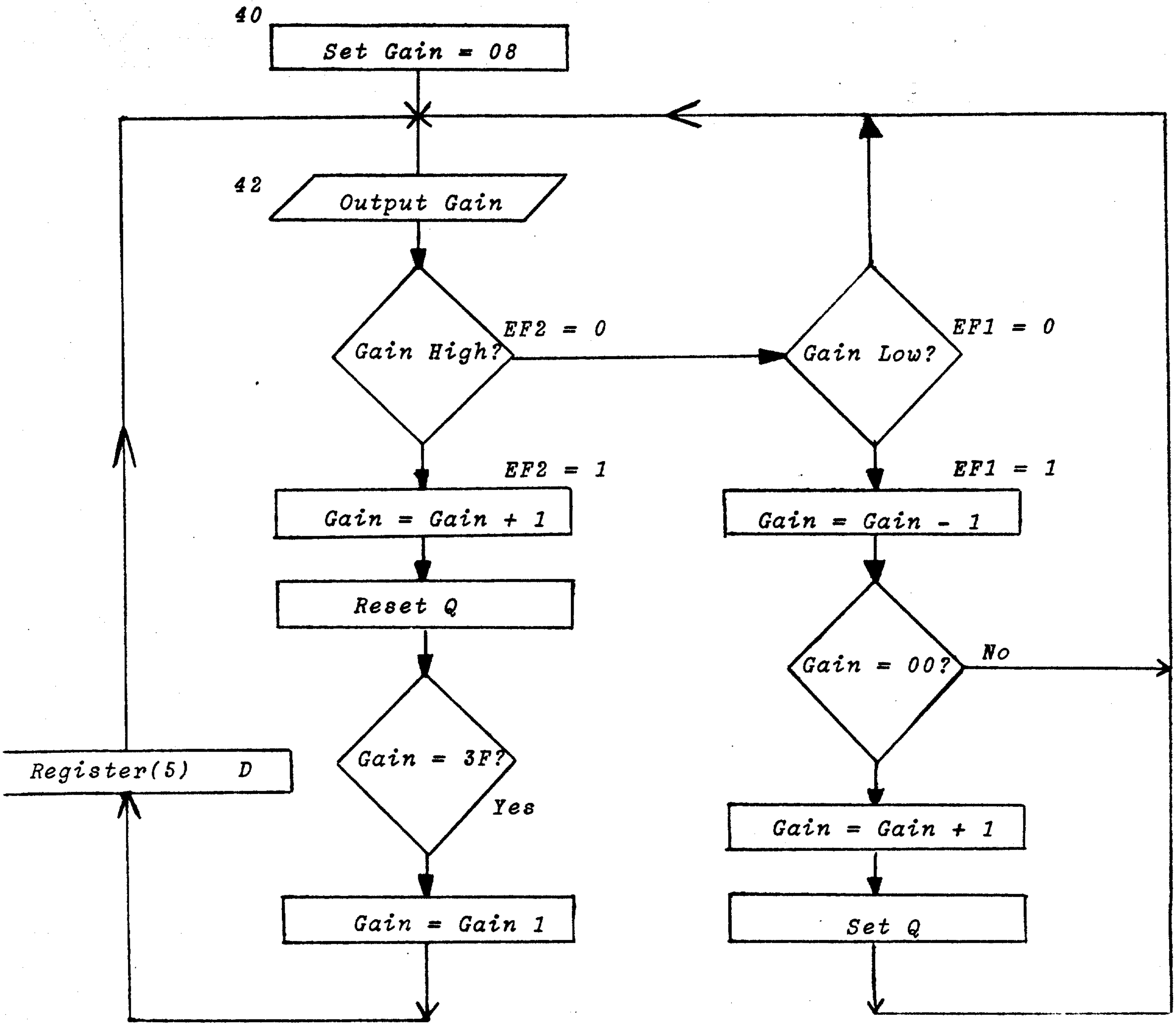


Figure 19 - HARDWARE FOR SOFTWARE DEVELOPMENT

Figure 20 - AUTOMATIC GAIN CONTROL - FLOW CHART 4/11/79



	EF1	EF2
Output Low	1	0
Output Normal	0	0
Output High	0	1

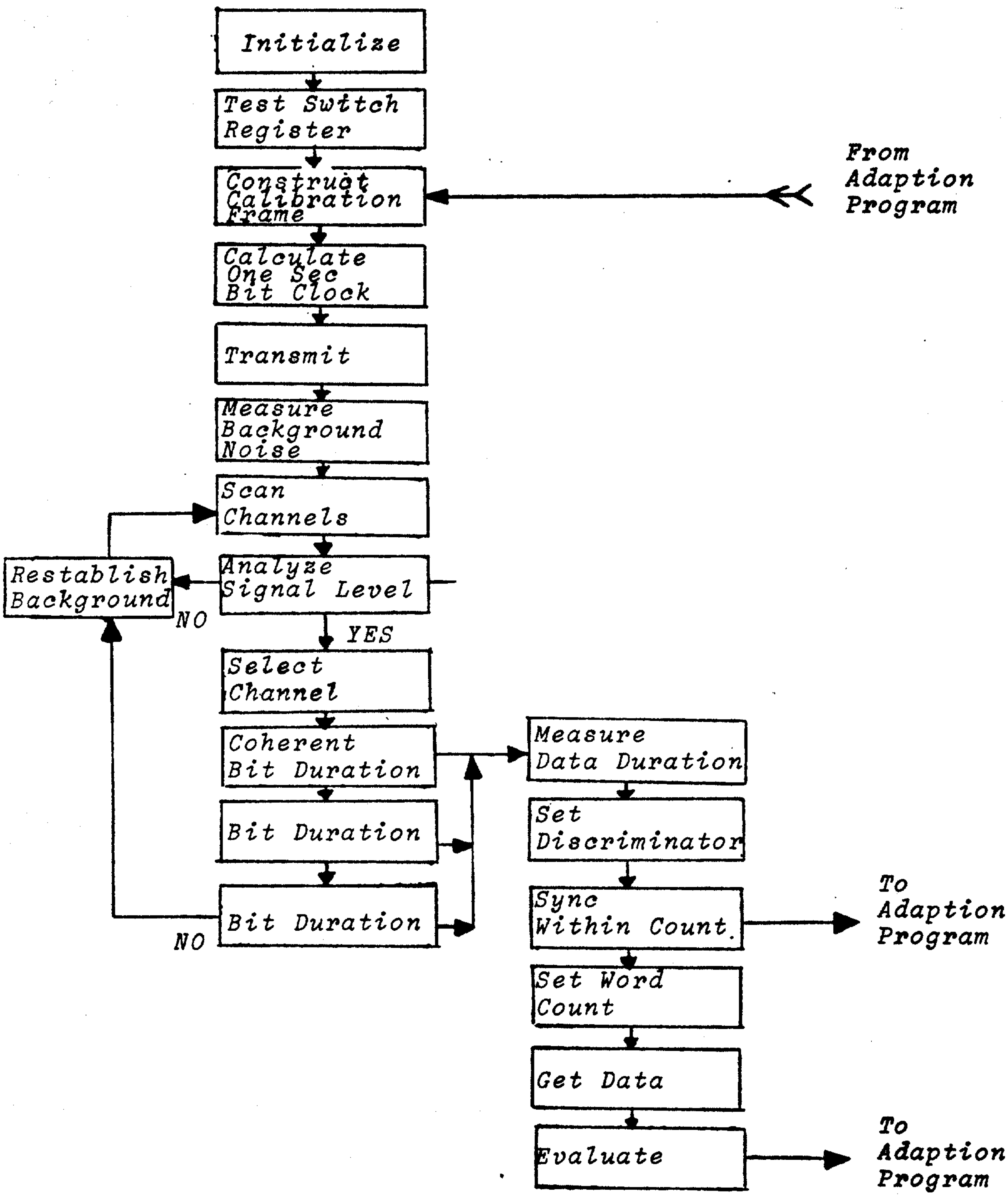


Figure 21 - TYPICAL RECEIVER/TRANSMITTER SOFTWARE

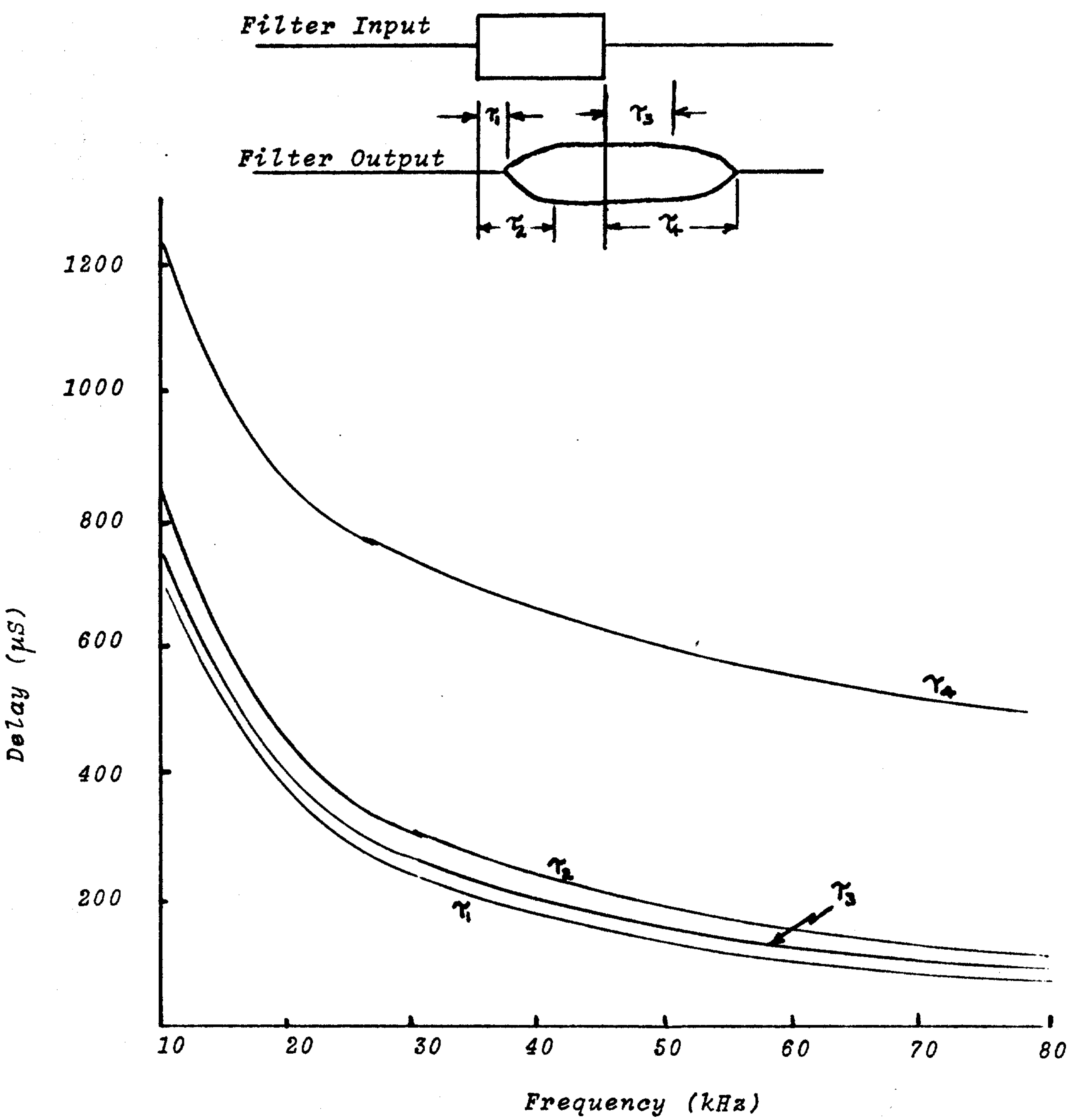


Figure 22 - DYNAMIC RESPONSE OF TRANSVERSE FILTER

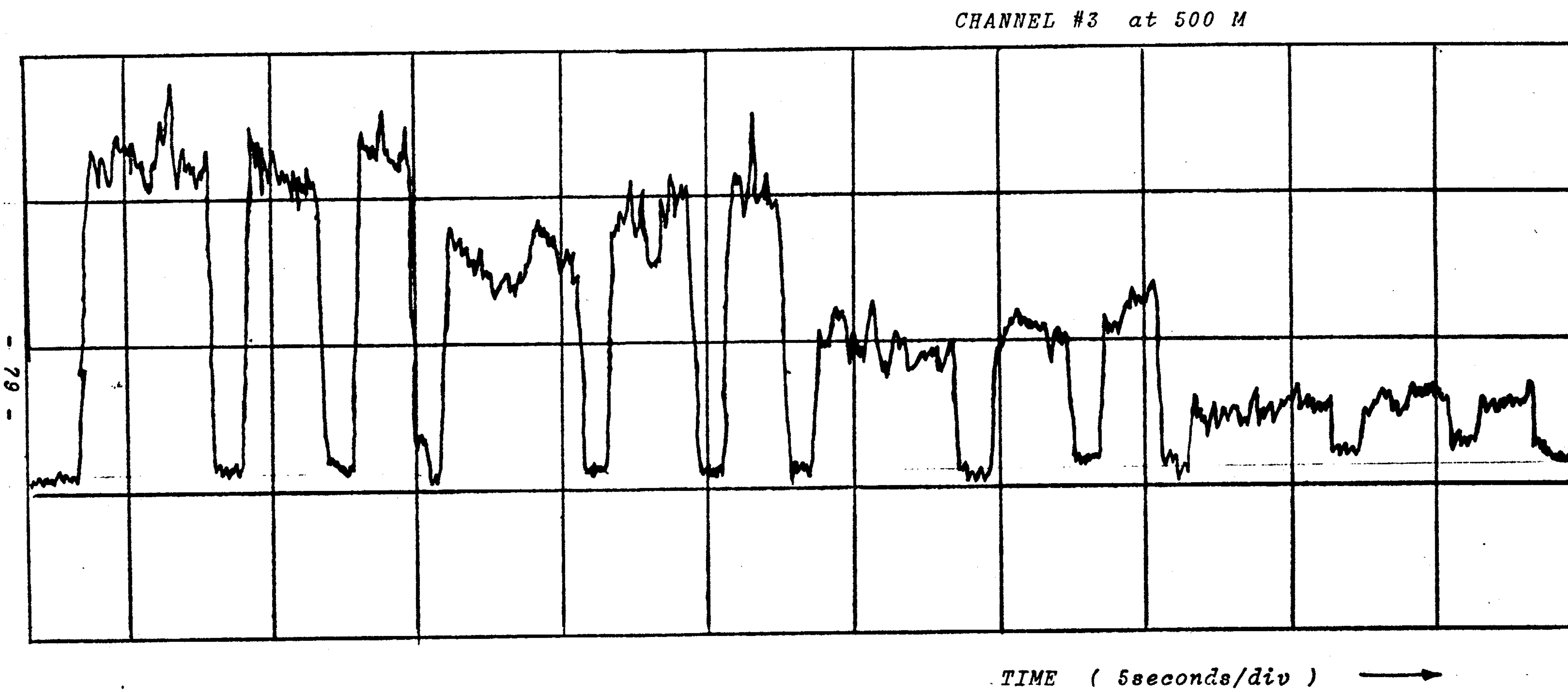


Figure 23. TYPICAL RECORDED AGC HISTORY
(Vertical scale - 10 db/div.)

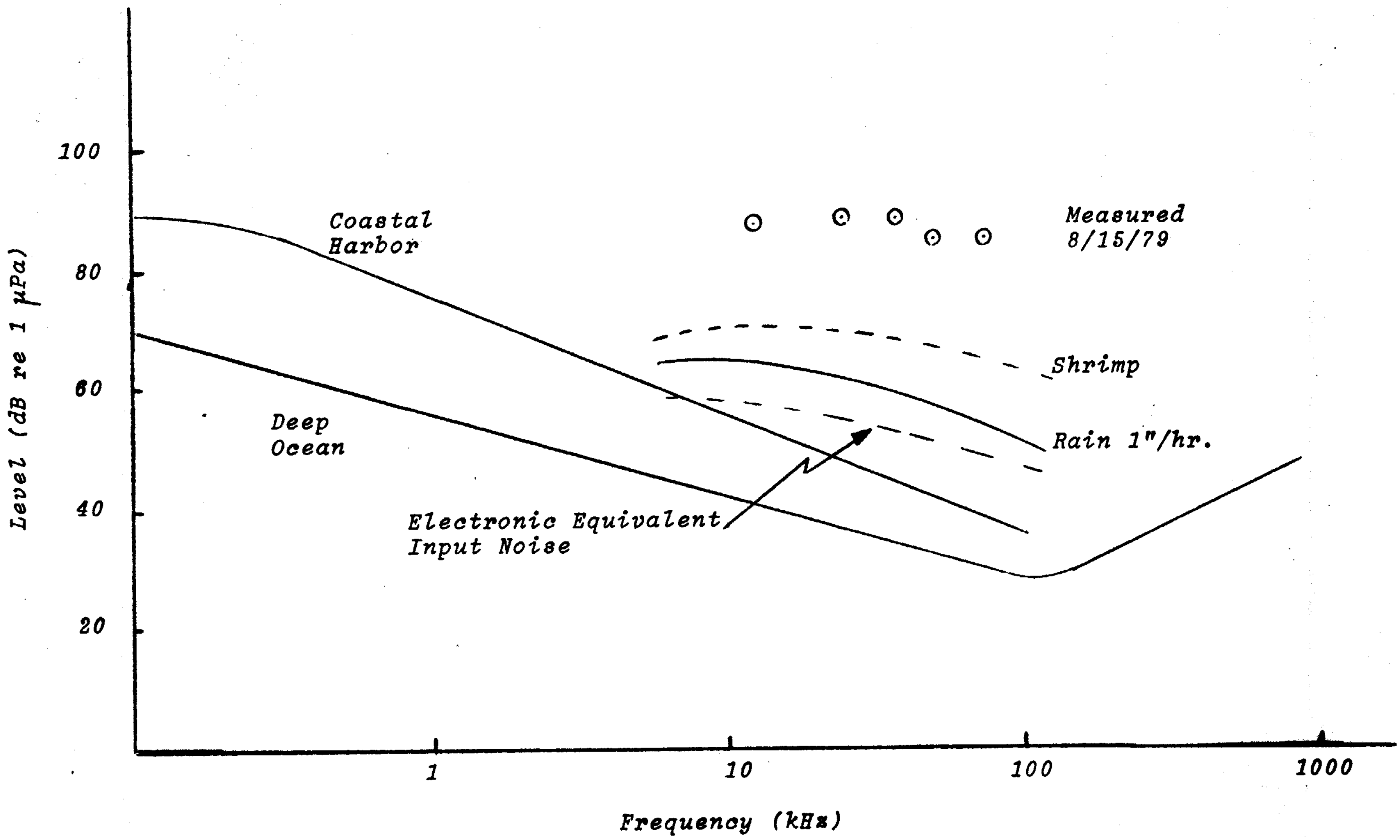


Figure 24 - SOME COMPARATIVE NOISE LEVELS

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